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Electrical and Electronic Engineering

Highlights

- Analysis of Power System
- Universal Hearing Aid System
- Three Phase Fault Analysis
- Premature Ventricular Contraction

Discovering Thoughts, Inventing Future

VOLUME 16 ISSUE 5 VERSION 1.0

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Three Phase Faults Analysis of Power System

By Tarang Thakur
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Abstract- The demand of electrical power is increasing day by day for households, agricultural, commercial, industry sectors etc. This paper is developed in order to maintain that electrical power required by these sectors, as in an electrical system, due to line to ground (L-G), line to line fault (L-L), three lines (LLL) various fault occurs. In this paper it has been discussed how to overcome this problem and for this a system is built, which has an auto reclosing mechanism of disconnecting the supply to avoid large scale damage to the control gears, load or manpower in the grid sub-stations. In this way a tripping mechanism is made in order to curb temporary and permanent faults.

Keywords: faults, power transformers, voltage regulator, relays, 555 timer.

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THREEPHASEFAULTSANALYSISOFPOWERSYSTEM

Strictly as per the compliance and regulations of :

Three Phase Faults Analysis of Power System

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I. Introduction

Current flows through all components of the electrical power system during normal operating conditions. There are various methods through which one can analyze any power system by calculating the system voltages & currents under normal & abnormal scenarios. Due to some unforeseen circumstances, faults could happen because of natural events or accidents like lightning strike, line to ground faults etc.

Three phase fault analysis and its protection mechanism main function is to ensure safety of equipments and maintain power system stability at high speed.

In order to protect the equipments of power system from faults, knowledge about system faults, their detection, and safe isolation of the faulted area is needed.

There are various types of fault. Some of them are Transient and Permanent Faults.

The occurrence of transient faults accounts to 70-90%. In overhead power lines, most of the faults are transient in nature. In a system comprising of various components like transformers, relays, Momentary tree contact, bird or other animal contact, Lightning Strike, Conductor Clashing or Insulator Flashover, Swinging wires and temporary contact can be the cause of transient faults. Thus, by de- energizing the line for short time transient faults can be cleared. Service to the line can be recreated by instant auto reclosing.[2][3]

The occurrence of Semi Permanent or Permanent faults abides to 10-30%. A semi-permanent fault can be effectuated when a small branch of tree falls on line. In such case of permanent fault, the fault can't be cleared by an instantaneous de-energizing of the line and subsequent auto reclosing. If there is a compeered time-delayed trip then system would let the branch to be burned away without any harm to the existing system. On an overhead line, a broken wire or conductor making a phase open, or a broken pole making the phases to short are the example of common and most often occurring permanent fault. Faults on underground cables are also the example of permanent fault. Most of the faults can be successfully cleared by using the appropriate tripping and auto reclosing mechanism. Proper tripping can de- energize the line for enough time period to pass the fault source and to de-energize the fault arc, then the system automatically recloses the line to maintain the power supply. Thus, auto reclosing mechanism can substantially decrease the outage time because of faults and gives a significant level of service consistency to the consumer and reliability of power system.

In the present scenario of power systems, automatic reclosing system has a very wide area where it can be applied.[2][3]

II. Types of Faults

a) Symmetric and Asymmetric Faults [3]

-Symmetric /Balanced Faults

These are very severe faults and occur infrequently in the power systems. These are of two types namely three lines to ground (L-L-L-G) and three lines (L-L-L). The occurrence of these faults is merely 2-5% in power systems.

- Asymmetric Faults/Unbalanced

These are very common as they occur way more time than symmetric faults and are less severe than former faults. These mainly constitutes of line to ground which is the most common fault (65-70%) , line to line (5-10%) and double line to ground (15-20%) faults.

In line to ground fault, a conductor makes contact with earth or ground. A line to line fault occurs when two conductors make contacts with each other mainly while swinging of lines due to winds. When two conductors make contact with ground then it a double line to ground faults.

b) Type of Faults on a Three Phase System [3]

A. L-to-G Fault (Line to Ground)
B. L-to-L Fault (Line to Line)
C. L-to-L-to-G Fault (Two lines to Ground)

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D. 3Line Fault (Three Phase)
E. 3L-to-G Fault (Three Phase to Ground)

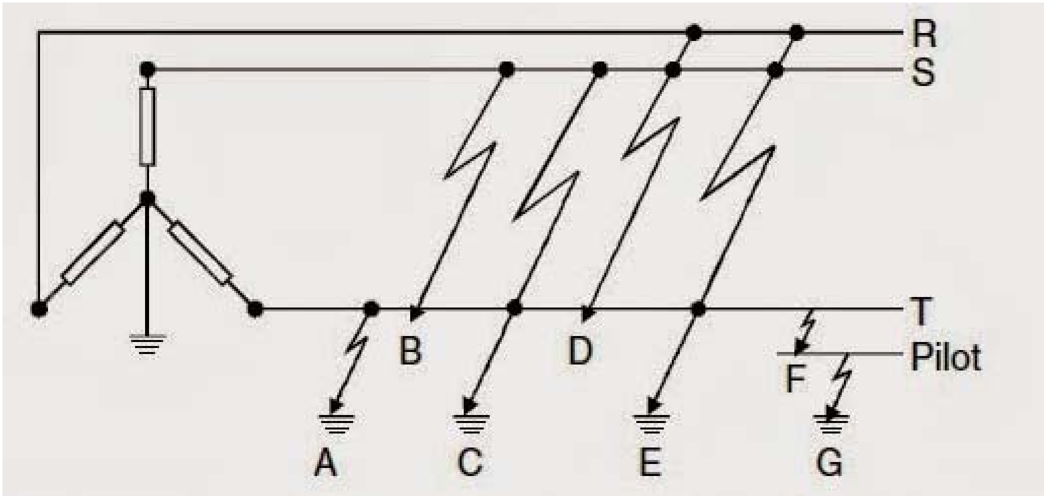

Figure 1 : Three Phase Faults [3]

III. Block Diagram

In figure 2, the basic arrangement of the implemented project can be found.

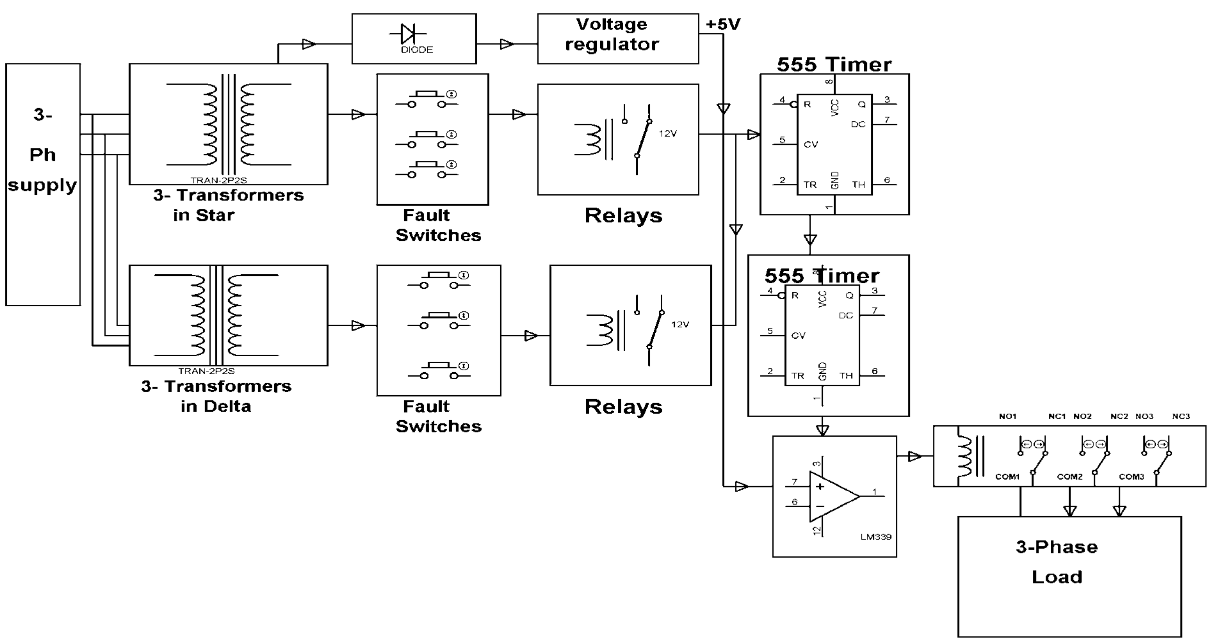

Figure 2 : Block Diagram Arrangement of the Project [1]

a) *Components Used*

The components required to establish the project, major of them are Power Transformers (Step Down), Voltage Regulator, Relays and 555 Timer, LM 358.[4]

i. *Power Transformers*

This is a equipment which is used to convert electricity from one alternating current (AC) voltage to another alternating current (AC) voltage with less loss of power. There are two types of transformers:-

-Step-up transformers that increase voltage.

-Step-down transformers that decrease voltage.

The ratio of number of turns in the primary & secondary windings determines that a transformer is step up or step down. If primary side windings are less than that of secondary sides then they are known as

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step up transformers. Vice versa they are step down transformers.

Figure 3 : 220v/12v Transformer

In this project transformers of (220v/12v) are used in groups of three each in which one group makes a star-star connection and the other group makes a star-delta connection.

ii. *Voltage Regulator (LM 7805)*

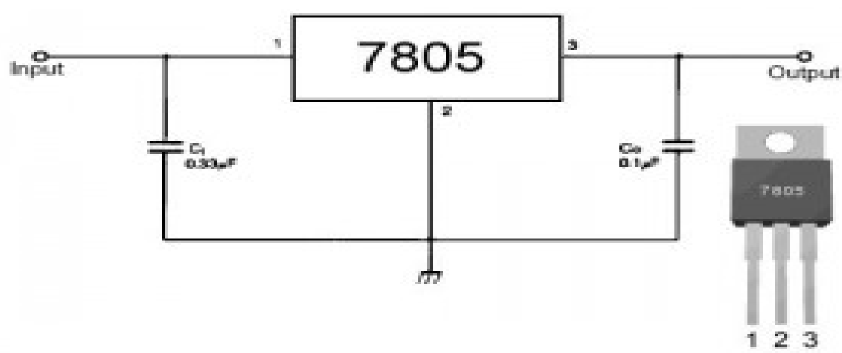


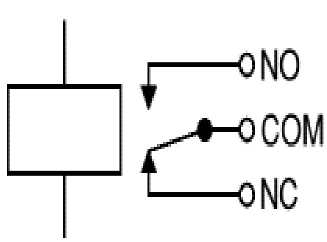
Figure 4 : Voltage Regulator

The LM78XX/LM78XXA is a series of three-terminal positive regulators are available in the TO-220/D-PAK package. It provides several fixed output voltages that helps in a wide range of applications. It helps in maintaining the constant voltage in the circuit as voltage from the voltage source is not constant. It retains thermal shutdown, internal current limiting and safe area protection so it makes them inextinguishable. Heat sinking is a major factor as it help them give over 1A output current and can provide a constant voltage of +5 volts. [1][2][3].

In Figure 4, LM 7805 has 3 pins denoted by 1-Input, 2-Ground and 3-Output.

iii. *Relays*

It is a device (electrically operated switch) which helps in protecting a system from severe damages by detecting and isolating faults on transmission and distributions lines by opening and closing of circuit breakers.



5.1

5.2

Figure 5

Here figure 5.1 shows the relay contacts. There are three terminal of the relay, one is common for the supply and the other two are normally open (NO) and normally close (NC).

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iv. *Integrated Circuit (ICs)*

An integrated circuit is a semiconductor chip on which thousands of resistor and capacitors are fabricated in order to function in particular way. They can be transistor, microprocessors, timer, counter, amplifier etc. ICs used here are:-

a. *555 Timer IC*

It is an integrated circuit, simple 8 pin Dual in Line package for timers, pulse generation and oscillator applications which can be used as Monostable and Astable Mode or as Timer IC. Its supply voltage is 3v to 15volt [3]

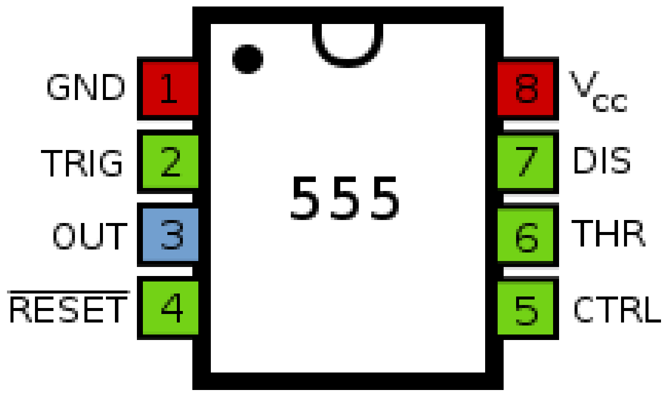

Figure 6 : 555 Timer

Pin	Name
1	GND
2	TRIG
3	OUT
4	RESET
5	CTRL
6	THR
7	DIS
8	V+, VCC

b. *LM 358 IC*

The LM358 series consists of two independent, high gain; internally frequency compensated operational amplifiers which were designed specifically to operate from a single power supply over a wide range of voltages. In this project, a potential divider is connected to its triggered input (Pin 2) and the output of 555 Timer is connected to the non inverting input (pin 3). It is acting like a comparator whose output is connected to 3 CO relay in order to drive it.

There are two cases:-

— If the inverting terminal input is greater than the non-inverting terminal input then the output of the comparator will be logic low (i.e. ground)
— If the inverting terminal input is less than non inverting input then the output of the comparator will be logic high and will drive the 3 CO Relay.

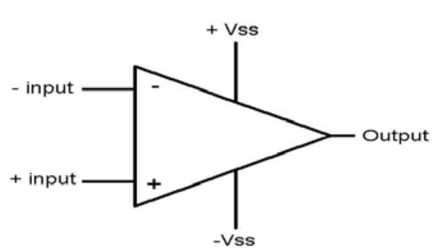

Figure 7 : LM 358

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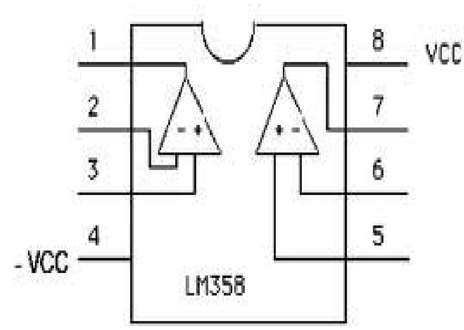
PIN CONNECTIONS

1 - Output 1
2 - Inverting input 1
3 - Non-inverting input 1
4 – VCC-
5 - Non-inverting input 2
6 - Inverting input 2
7 - Output 2
8 _ VCC+



b) Quantities of components used

Main components used	Quantity
Power Transformers (12v,500mA	6
Voltage Regulator	1
Relays (12v)	6
3C/O Relay	1
555 Timer	2
Operational Amplifier (LM 358)	1
Transistor (Bc547)	6
2 pin push buttons	6
LEDs	10
Diodes (IN 4007)	15

c) Voltage at IC Pins

Integrated Chips (IC)	Pin (No.)	Voltages at pins without IC (Voltage)	Voltages at pin with IC (Voltage)
555 Timer (Monostable Mode)	5	0	11
	6	16	0
	7	16	0
555 Timer	1	5	0
(Astable Mode)	6	5	0
	7	5	0
LM 358 (Operational Amplifier)	3	0	3

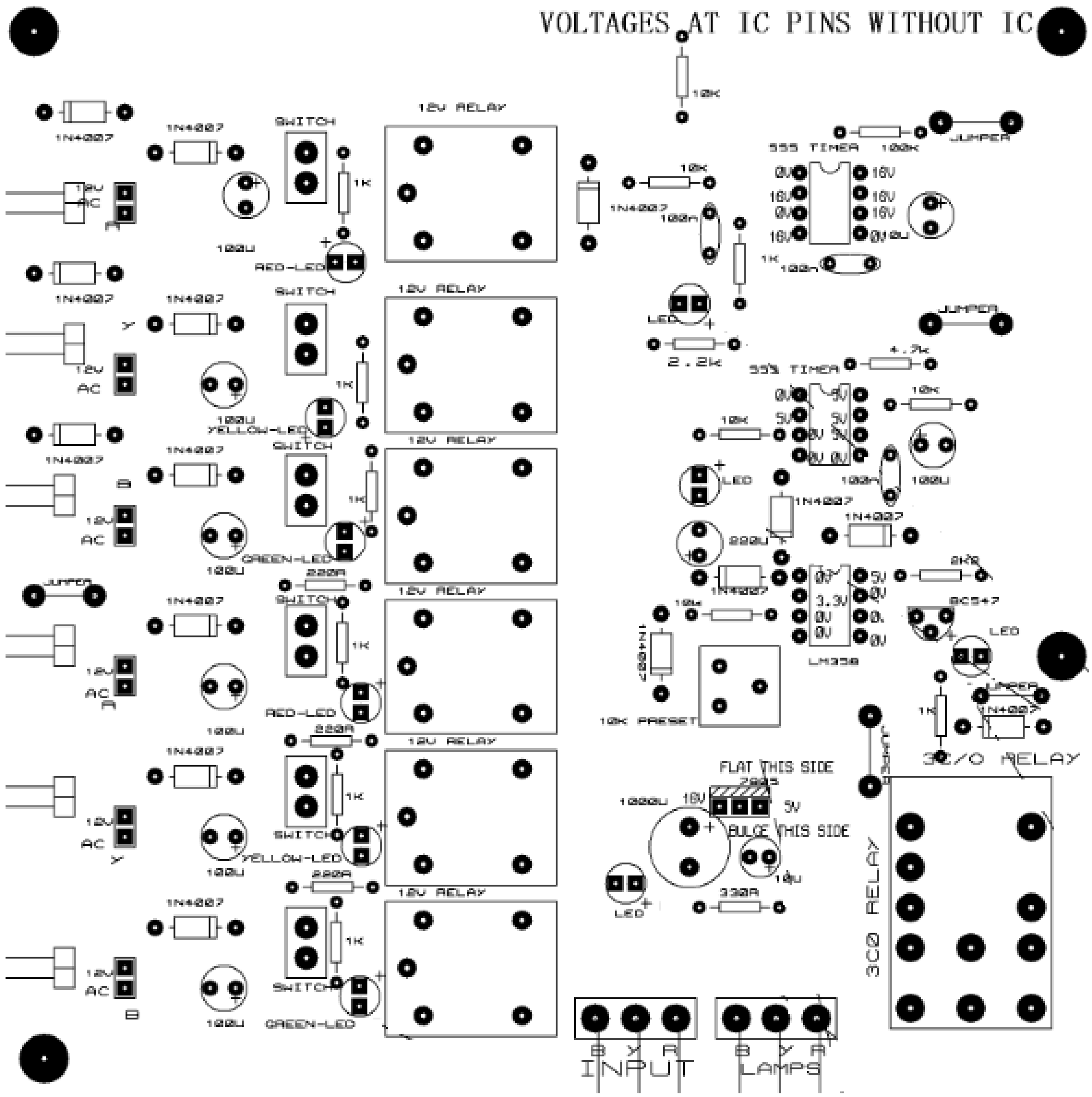


Figure 8 : Voltages at IC Pins without IC (PCB)

Source: - http://www.edgefxkits.com/

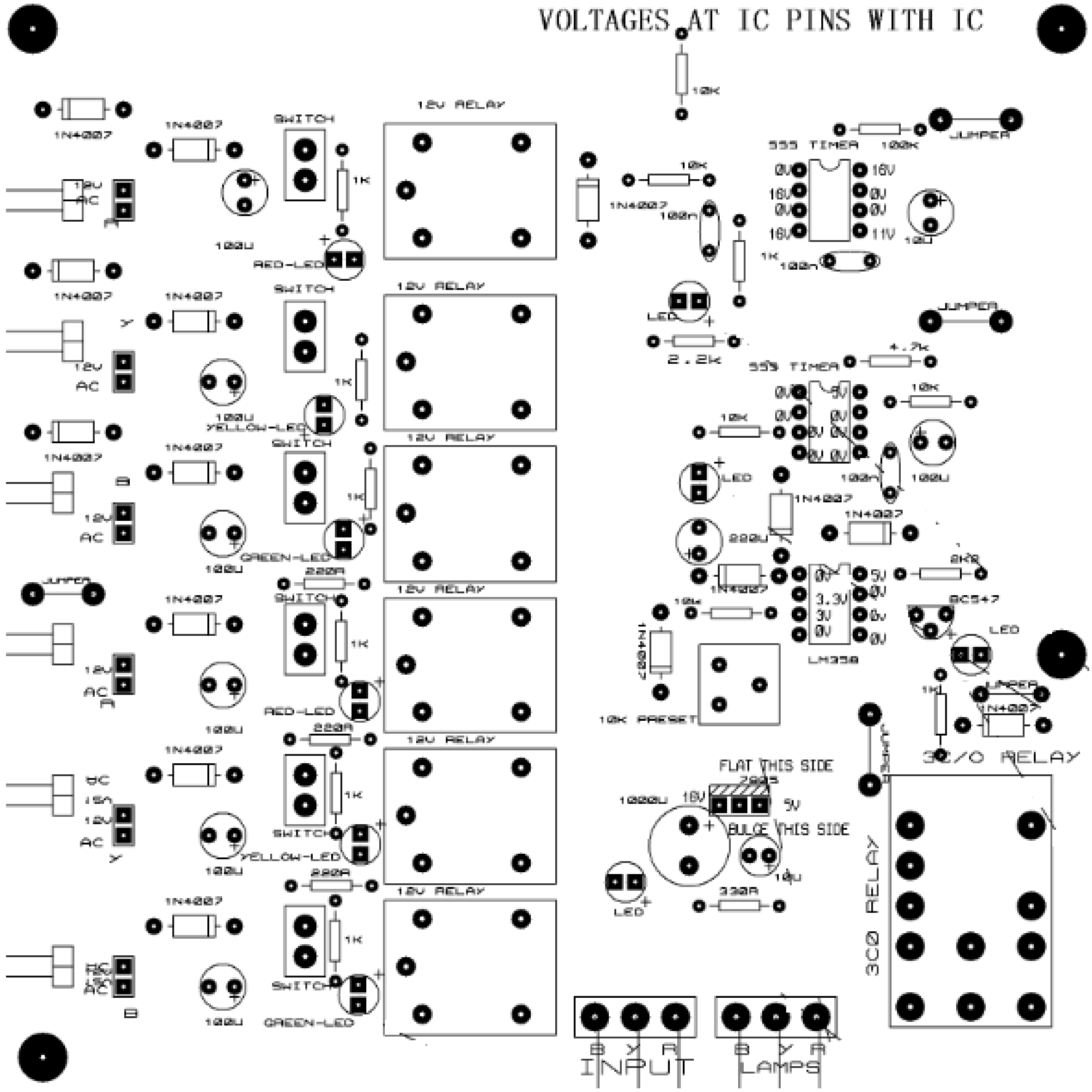


Figure 9 : Voltages at IC Pins with IC (PCB)

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IV. Simulation

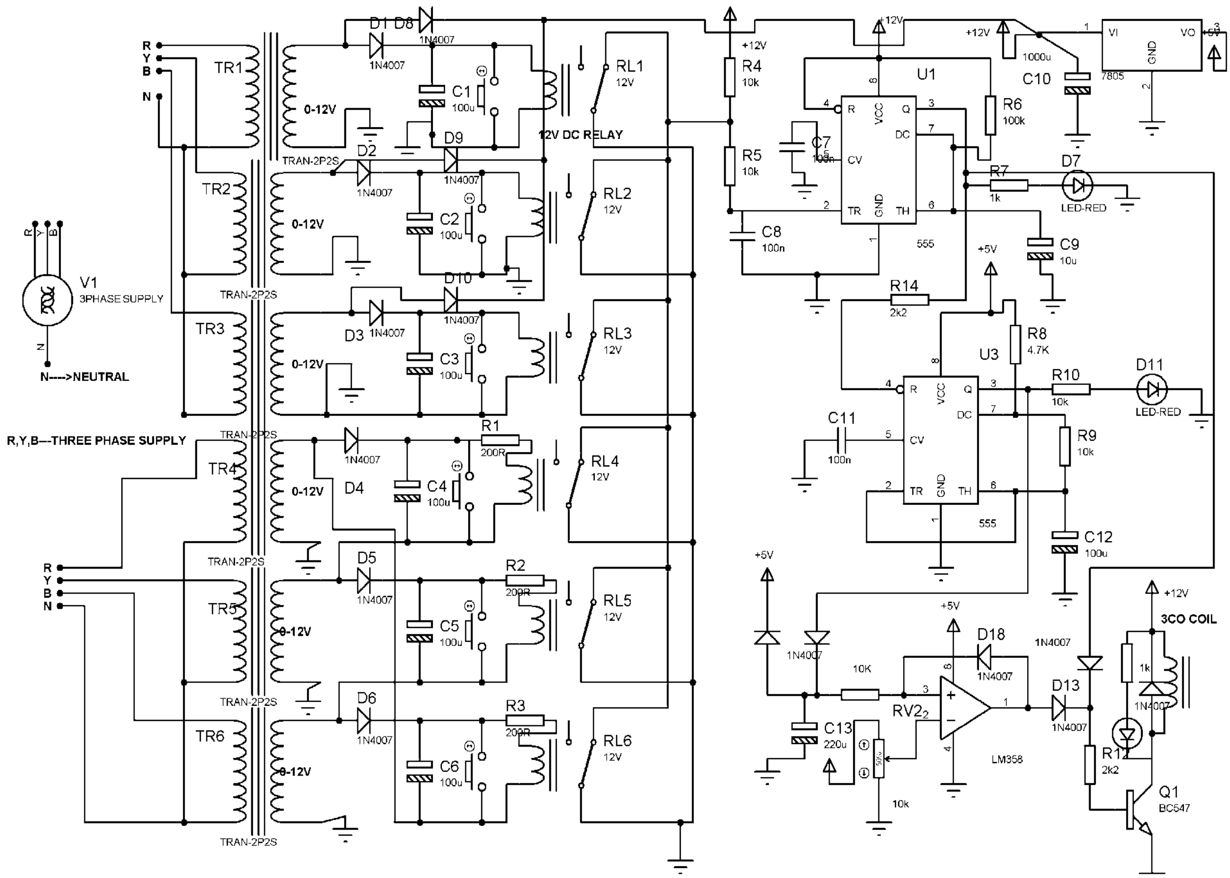

Figure 10 : Circuit Simulation [5]

The initial stage, the circuits have been designed and simulated in PROTEUS. After this all the faults were produced like line to ground (L-G), line to line (L-L), double line to ground (2L-G), 3 line to ground (3L-G). The circuit diagram can be found in Figure 9.

a) *Operational Explanation*

i. *Working*

Six step down transformers are connected to the board producing 12v to the circuit. These six are divided in group of two each, in which one group is connected in star-star connection and the latter is connected in star-delta connection. All the six transformers are the rectified and filtered individually with the help of rectifying circuit and are then given to corresponding relay coils.

Push buttons are connected to each relay coil to create fault conditions. The Normally Closed (NC) contacts of all the relays are then made parallel while all common points are grounded. The parallel connected point of NC is then linked to Pin 2 of 555 Timer (Monostable Mode) through a resistor R5. The output (Pin 3) of the same timer U1 is connected to Reset (Pin4) of the other 555 timer (Astable Mode). LED is connected at their output each to indicate their status. The output (Pin 1) of 555 Timer (U3) is given to Op-Amp LM 358 through wire 11 and d12 (IN4007) to the Non-Inverting Input (Pin 3) which acts as a comparator. It compares the value of Pin 2 (Inverting Input) and 3 (Non-Inverting Input) of LM358. The voltage of Pin 2 is kept at fixed/constant voltage with the help of a Potential Divider. It is generally kept higher than the Pin3 of Operational Amplifier so that Pin 1 ie Output of LM 358 develops low (Zero Logic) which fails to operate 3 CO relay through the transistor Q1 and the same is used for disconnecting the load used in fault conditions.[1]

ii. *Operational Procedure*

Transformers and Lamp Bulbs are connected along with Three Phase Power Supply (230V). After the board is powered by 3 phase supply, all relay coils get DC voltage and due to this the common points disconnects from NC contacts and moves to the NO contacts. When push buttons are pressed, it disconnects the relay and due to this the common points moves to the NC position to provide a logic low at a trigger pin (Pin 2) and the output (Pin 3) which is linked to reset pin (Pin 4) develops high logic indicated by D11 flashing LED of 555 Timer (u3) which is in Astable Mode.

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-If fault is temporary

If any push button is released after a short time, 555 Timer (U1) in Monostable Mode disables U3 due to which the output of U3 goes to zero.

-If fault is permanent

If any push button is pressed for a longer duration, then the output of 555 Timer (U3) present in Monostable Mode provides a longer duration of active situation for 555 Timer (U3), output of the same charges the capacitor C13 through R11. The output (Pin 1) of Operational Amplifier (LM 358), though acting like a comparator gets high which in turn drives the 3 CO relay through transistor Q1 to switch off 3 phase load (lamps in this case).[1]

V. Hardware Implementation

Step 1

First, a circuit is printed on Printed Circuit Board (PCB).

Figure11 : PCB

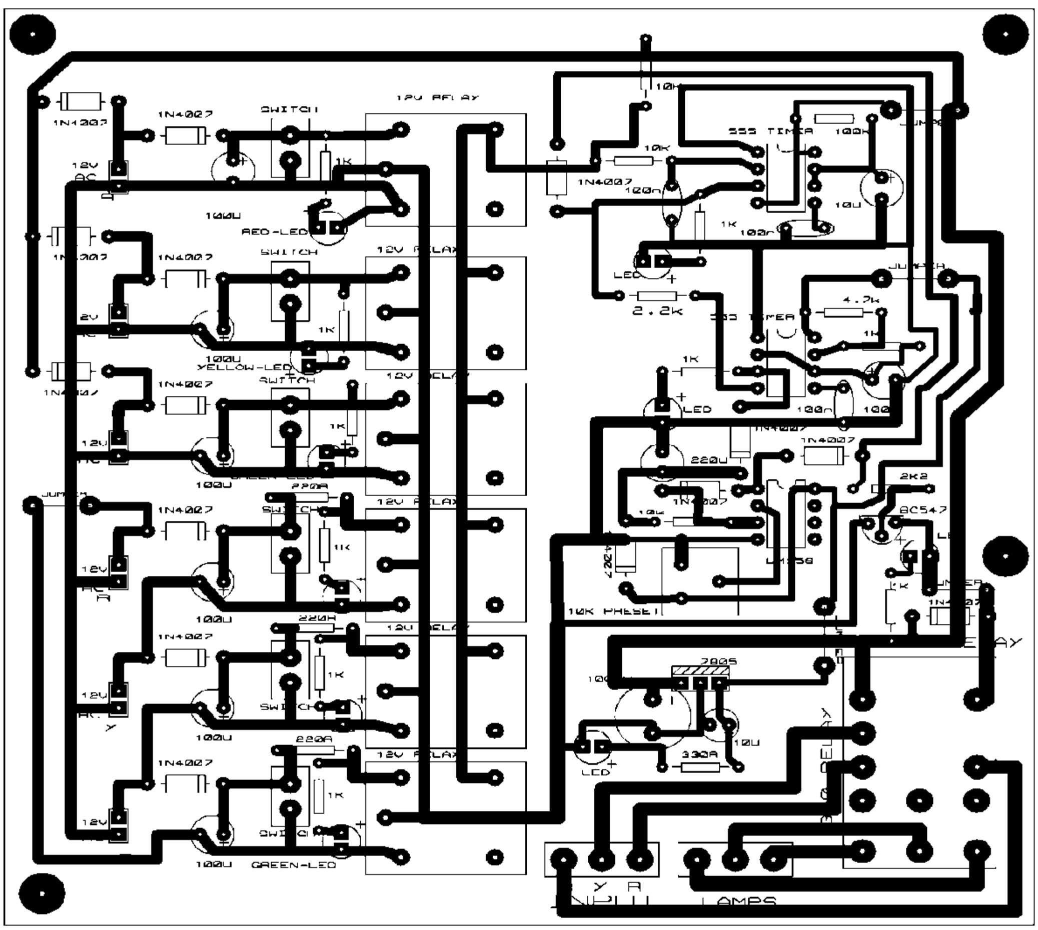

Figure 12 : PCB layout design [5]

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All the components are then adjusted and soldered to the board carefully. After this, secondary side of six transformers is connected to the relays of the circuit designed on the PCB in order to produce faults. In the groups of three each, one group is connected in star-star connection and the other group is connected as star-delta connection. After this 6 lamps are connected along with 230 V supply.

Now all the circuit should be fitted on wooden board with help of drilling.

Figure 13 : Wooden Board

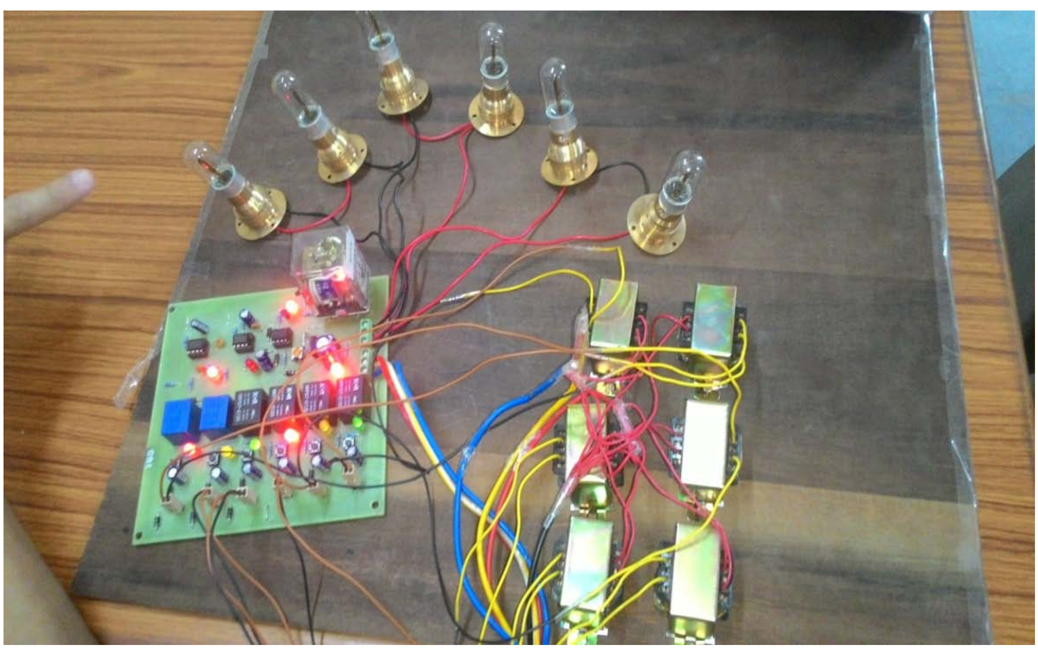

Figure 14 : Implementation of Hardware Model

Step 2 Powering the circuit

Process of starting the circuit-

- Turn the preset fully anti-clockwise and connect six transformers to the board.
- Use variac so that there is no sudden over voltage in the system. Slowly rotate it clockwise so that system starts properly.
- All 6 relay Led, power supply Led and 3 c/o Relay shall be glowing.

- Both LEDs of 555 Timer shall be in off position.
- Turn the preset slowly clockwise so that 3 C/O Relay led goes off.

That completes the setting procedure and the board is ready for use.

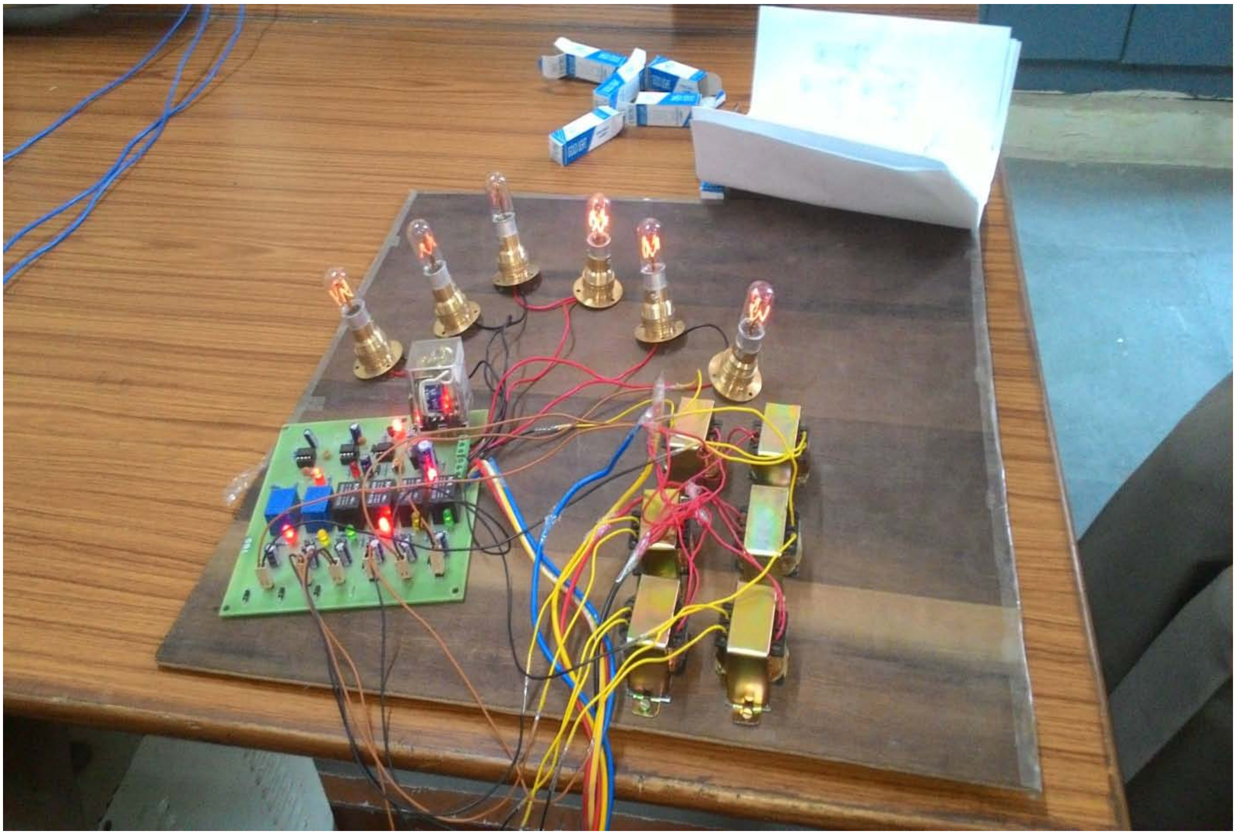

Figure 15 : Powering the circuit

VI. Conclusion

This project is designed to develop an automatic tripping mechanism in case of temporary and permanent faults. A schematic diagram has been developed by Proteus 8.3 software which helps in simulation of different types of faults which in turns helps to build hardware for the project. In this case, two 555 timers are used along with relays in order to determine whether fault is temporary or permanent.[3] Nowadays, a mechanism to send message to the authorities via SMS by interfacing a GSM modem GPRS based network is used for tracking transformers. In future there can be more advancement in Three Phase Fault Analysis System like:-

— GPRS based network is used for tracking transformers.

— A mechanism to send message to the authorities via SMS by interfacing a GSM modem

— Improvements to human machine interface

— Long Distance Data Transmission [4]

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Non-Uniform Chebyshev Excitation of a Linear Broadside Antenna Array Operating at 1.8GHz

By Aras Saeed Mahmood & Dlnya Aziz Ibrahim

University of Sulaimani

Abstract- A given configuration of a non-uniform antenna array for which the elements are equally spaced with unequal amplitude excitation using chebyshev excitation method has been studied in this work. The impact of the number, spacing and amplitude excitations of the elements on the radiation characteristics has been analyzed and compeared with its analogues of a linear broadside uniform and non-uniform binomial antenna arrays using 4NEC2 simulation software. The spacing between the elements varied from 0.1λ to 2λ in steps of 0.02λ for a number of elements up to 10 elements. For GSM application at 1.8GHz frequency the best configuration to achieve the required gain of 13.9dBi was 7-elements at 0.86λ spacing.

Keywords: chebyshev array, gain, half power beamwidth, max. side lobe level, no. of side lobes and radiation pattern.

GJRE-F Classification : FOR Code: 100501

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Aras Saeed Mahmood [α] & Dlnya Aziz Ibrahim [σ]

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Keywords: chebyshev array, gain, half power beamwidth, max. side lobe level, no. of side lobes and radiation pattern.

I. Introduction

Wireless communication has become an integral part for modern word. The most popular standards for mobile phones in today's word is Global System for Mobile communication (GSM). A 4x4 rectangular microstrip patch antenna with the gain about $(13.8 \sim 14.4)$ dBi for GSM application was presented by [1]. [2] achived the same gain range using a 2x2 microstrip patch antenna for the same application. A 13.7dBi gain for the same application was achived by [3] using a 7 dipole elements with 0.82λ spacing forming a uniform linear broadside array operating at 1.8GHz. An anaolugous binomial excitation non-uniform linear broadside dipole antenna array has been used by [4] to achieved a 12.8 dBi gain with a 10-element and 0.82λ spacing. In this study another technique for current excition known as Chebyshev excition of a non-uniform linear dipole antenna array has been studied for GSM application at 1.8GHz. Linear array antenna has a wide range of applications in radar and communication systems due to higher directivity, low side lobe and high gain when compared with other kinds of single radiating element antenna [5].

Dolph-Chebyshev arrays are typical examples of non-uniform arrays [6]. The excitation coefficients for this array relates to Tschebyscheff polynomials. A Dolph-Tschebyscheff array with no side lobes (or side lobes of $-\infty$ dB) reduces to the binomial design [7]. Tschebyscheff polynomial is defined by equation:

$$T_m(x) = \cos(m \cos^{-1} x) \quad for \ -1 \le x \le +1 \quad (1)$$

$$T_m(x) = \cos h(m \cosh^{-1} x) \quad for \ \ x < -1, x > +1 \quad (2)$$

Where T denotes the Tschebyscheff and m is the order of the polynomial. For higher terms can be had from the recursion formula:

$$T_{m+1}(x) = 2\,x\,T_m(x) - T_{m-1}(x) \qquad (3)$$

Steps to be followed while calculating Dolph-Tchebyscheff amplitude distribution are given by [8] which give Dolph- Tchebyscheff optimum distribution for a specified side lobe level.

II. Design and Simulation Results

In this work some basic radiation characteristics for a non-uniform linear dipole antenna array has been analized through the variation of the number of the elements and the spacing between them using Chebyshev method for specifing the amplitudes of the excitation currents of the elements. For a 1.8 GHz operating frequency the length (L) and the radious (R) of the array element (half wave length dipole antenna) has been calculated from [9,10] and the elements were arranged parallel to each other along the Z-axis. The results were also compaired with that of a similar uniform and binomial arrays.

a) *Effect of Number of Elements on the Performance of Dipole Array Antenna*

The first proposed study was the impact of the number of elements on the radiation charactistics for linear array chybeshiv excitation. When the spacing between the elements were fixed at (0.5λ) and the number of elements were changed from 2 to 10 elements the current excitation amplitudes for the elements has been calculated using the steps given by [8] using the major to minor lobe ratio of 20dBi. Table (1) tabulates the current excitation amplitudes for all the 10- elements.

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Table 1 : Excitation coefficients for Dolph Tschebyscheff for different number of elements

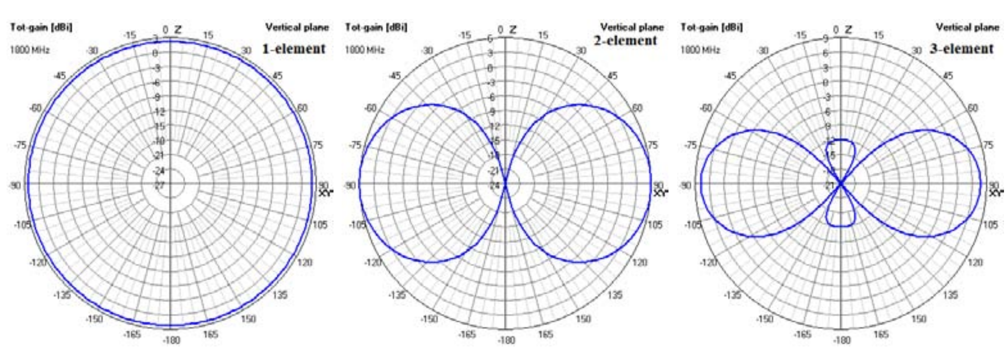
No. of Elements										
N=1	1									
N=2	1	1								
N=3	1	1.636	1							
N=4	1	1.736	1.736	1						
N=5	1	1.607	1.929	1.607	1					
N=6	1	1.439	1.855	1.855	1.439	1				
N=7	1	1.276	1.683	1.837	1.683	1.276	1			
N=8	1	1.139	1.507	1.72	1.72	1.507	1.139	1		
N=9	1	1.023	1.355	1.596	1.662	1.596	1.355	1.023	1	
N=10	1	0.926	1.212	1.436	1.558	1.558	1.436	1.212	0.926	1

The design simulation and optimization processes were carried out with the aid of the 4NEC2 simulator (antenna design procedure using 4NEC2 were mentioned in [11]) after equating the phase of the elements current to zero for broadside array. Table (2) tabulates some of the outputs of the 4NEC2 simulator for this section.

Table 2 : Gain, HPBW, SLL max and No. of SLLversus the number of elements

No. of elements	Gain (dBi)	HPBW (vertical plane) degree	HPBW (horizontal Plane) degree	SLL max. (dBi)	No. of SLL
1	2.14	360	80	-∞	0
2	5.97	60	80	-∞	0
3	7.72	40	80	-12.1	2
4	9.03	28	80	-10.9	4
5	10.07	24	80	-9.28	6
6	10.92	20	80	-8.84	8
7	11.62	16	80	-7.54	10
8	12.23	12	80	-7.4	12
9	12.75	12	80	-6.53	14
10	13.22	12	80	-6.37	16

The above data (Table 2) was translated to Fig. (1) through the pattern representation where the horizontal HPBW remained unchanged while the vertical one decreased with the elements up to 8 elements then kept constant. The number of side lobes, max. SLL and the gain all increase with the number of elements. Fig. (1) shows all these variations.



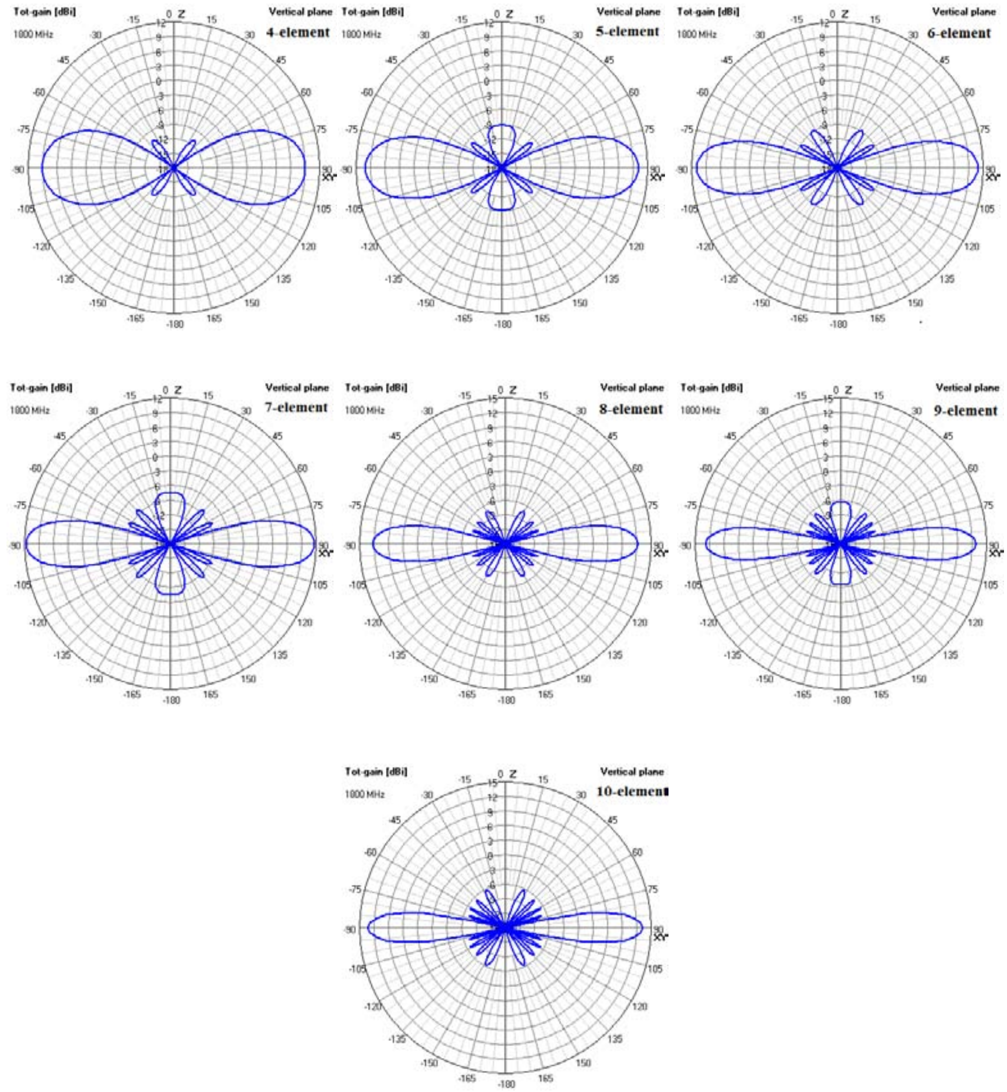


Figure 1 : Radiation pattern in vertical plane vs. the number of the elements

b) Effect of Element Spacing on the Performance of the Dipole Antenna Array

The second part is to investigate the impact of the variation of the spacing of a chebyshev excitation linear dipole array with 10 elements has been calculate using the steps of [8] to calculate the excitation current amplitudes. Table (3) contains some of the outputs of the utilized software (4NEC2); it shows a smooth and a systematic increase of both the gain, max.SLL and the number of side lobes respectively with the spacing up to 0.9λ while the HPBW in vertical plane decreases with the spacing and that for horizontal plane remained constant.

Table 3 : Gain, HPBW, SLL max and No. of SLL versus the spacing for 10 element array

Elements spacing(λ)	Gain (dBi)	HPBW (vertical Plane) degree	HPBW (horizontal Plane) degree	max. SLL (dBi)	No. of SLL
0.1	5.89	60	80	-∞	0
0.2	9.33	28	80	-10.8	4
0.3	11.1	20	80	-8.75	8
0.4	12.31	12	80	-7.52	12
0.5	13.22	12	80	-6.37	16
0.6	13.96	8	80	-5.93	20

0.7	14.58	8	80	-5.37	24
0.8	15.12	8	80	-4.97	28
0.9	15.58	4	80	-0.15	34
1	11.30	4	76	-8.45	28

Figure 2 : Illustrates the radiation patterns for a 10 element array at different spacing starting from 0.1λ to one λ

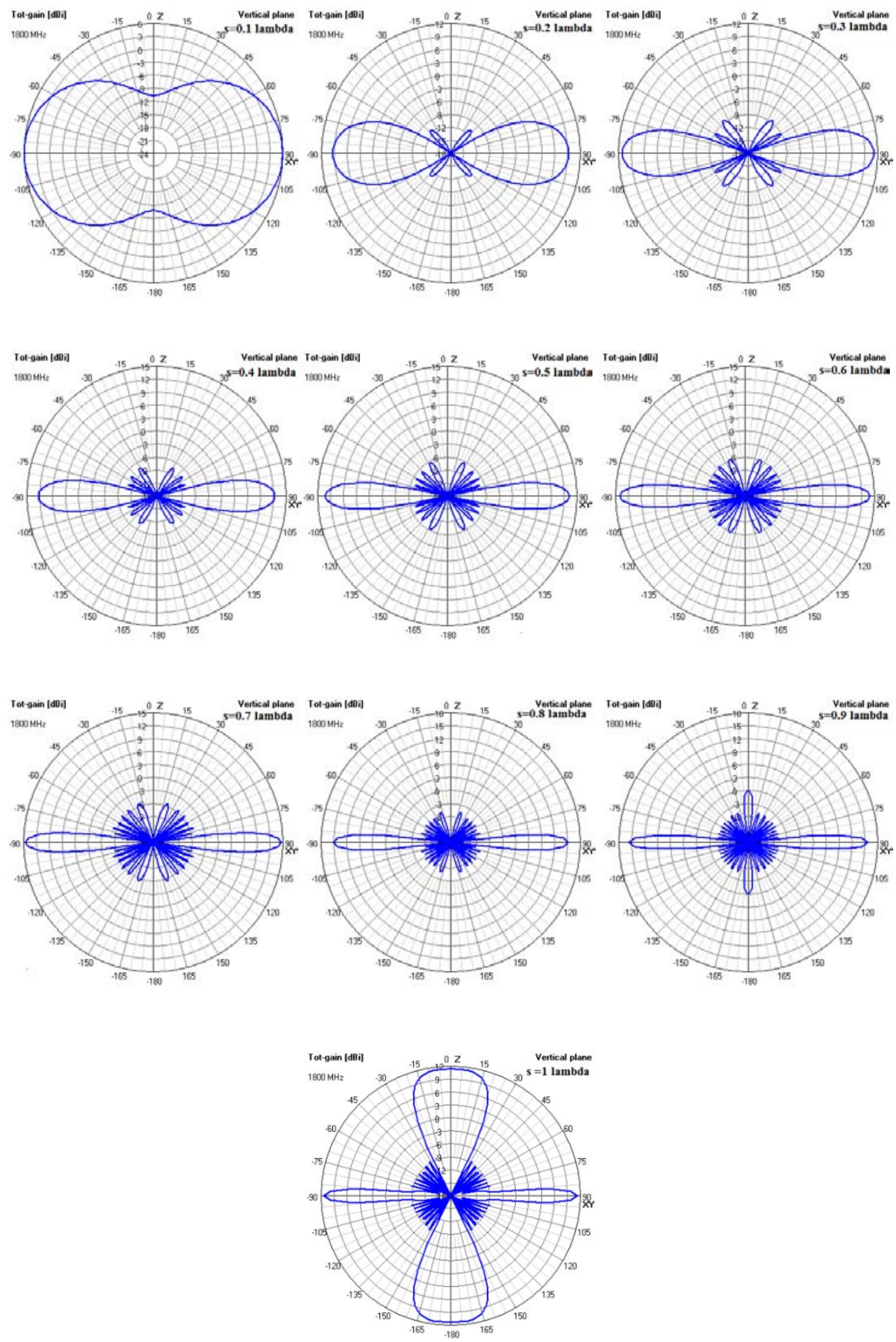


Figure 2 : Radiation pattern for a 10 elements array at different spacing

c) Optimization for Best Spacing for the Dipole Antenna Array

At the third part of this analysis, both the number of the elements and the spacing between them were varied to see their impacts on the same radiation characteristics as before . The spacing is varied up to 2λ at steps of 0.02λ for each number of the elements from 2 to 10 elements separately. Table (4) tabulates the outputs for the specified parameters at the best spacing for maximum gain only.

Table 4 : HPBW, Gain, max. SLL, no. of SLL for the max. gain at different no. of elements

No. of element	Element spacing (λ)	Gain (dBi)	HPBW (vertical plane) (degree)	HPBW (horizontal plane) (degree)	max. SLL (dBi)	No. of SLL
2	0.66	7	44	80	0.73	2
3	0.72	9.20	28	80	0.23	6
4	0.75	10.8	20	80	-3.1	10
5	0.8	12.1	16	80	-2.04	14
6	0.84	13.1	12	80	0.26	18
7	0.86	13.9	10	80	-0.12	22
8	0.86	14.5	8	80	-5.63	26
9	0.88	15.1	8	80	-5.12	30
10	0.9	15.6	6	80	-0.15	34

The variation of the gain with the spacing up to 2λ for different number of array elements is shown in the Figure 3 below.

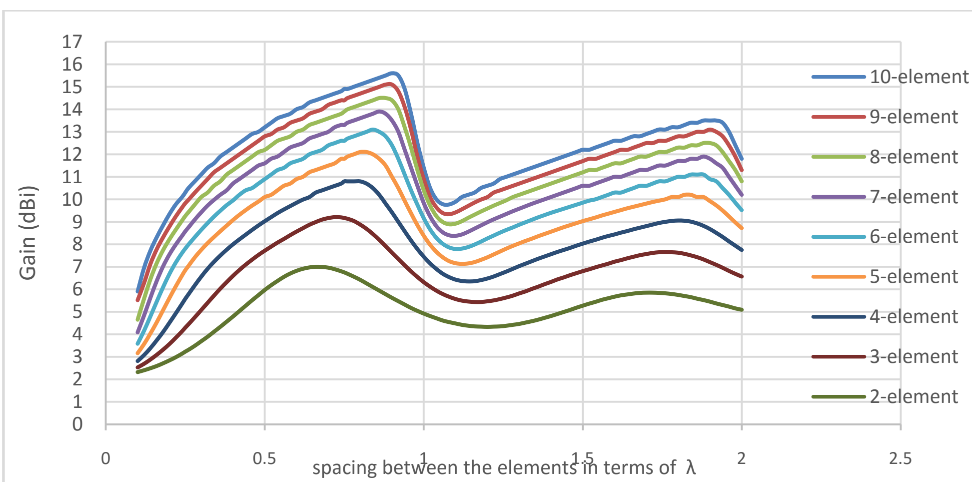


Figure 3 : Gain vs. the spacing for different number of elements

The above data (table 4), clearly shows the maximum gain of the different number of elements produced at different spacing between them; the level of side lobe is changing from one to another and the number of side lobes increases when the number of elements increases; for increasing one element, four side lobes add to the radiation pattern. The same results are sumarized in Fig. (4) throught the radiation pattern of each array with the specifications of Table (4).

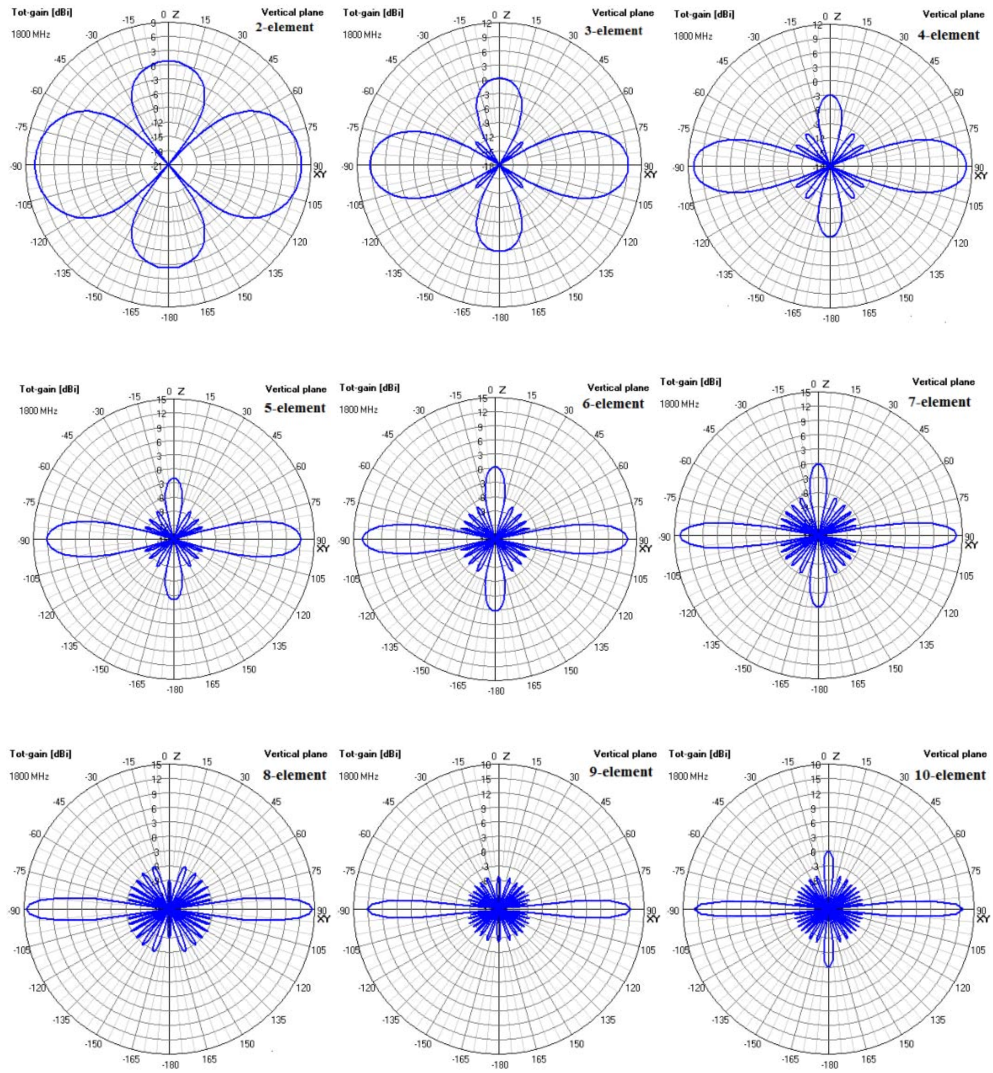


Figure 4 : Radiation patterns for different number of elements and spacing at maximum gain (best spacing)

III. Comparison between Different Types of Excitations Arrays

For the half wavelength dipole antenna array operating at 1.8GHz the radiation characteristics of the chebyshev excitations have been compaired with the same correspounding uniform array of ref.[12] and the binomial excitation of ref. [research 2]. Table (5) tabulates the variation of these characteristics with the number of the elements having 0.5λ spacing between them.

Table 5 : Variation of the Gain, HPBW, max. SLL and No. of SLL of different excition arrays with the number of elements

Number of Elements	uniform Array ref.3				Binomial Array ref.4				Chebyshev Array			
	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL
1	2.14	80	-∞	0	2.14	80	-∞	0	2.14	80	-∞	0
2	5.97	60	-∞	0	5.97	60	-∞	0	5.97	60	-∞	0
3	7.80	36	-1.3	2	7.41	44	-∞	0	7.72	40	-12.1	2
4	9.2	28	-1.97	4	8.29	36	-∞	0	9.03	28	-10.9	4
5	10.2	20	-1.82	6	8.92	32	-∞	0	10.07	24	-9.28	6

6	11.06	16	-1.36	8	9.42	28	-∞	0	10.92	20	-8.84	8
7	11.74	16	-0.89	10	9.82	24	-∞	0	11.62	16	-7.54	10
8	12.35	12	-0.61	12	10.2	24	-∞	0	12.23	12	-7.4	12
9	12.88	12	-0.07	14	10.4	20	-∞	0	12.75	12	-6.53	14
10	13.36	8	0.29	16	10.7	20	-∞	0	13.22	12	-6.37	16

From the above table (5) it is clear that for all the arrays as with different elements both uniform and cebyshev excitations give almost the same gain and it is more than that of the binomial arrays. Ingeneral uniform array gives more directive beam (narrow HPBW) while binomial arrays give wider major lobes.

At 0.5λ spacing the binomial array has not any side lobes while both uniform and chebyshev arrays have the same number of side lobes for any element numbers but with different intensities such that the intensity of the side lobes for the uniform array excitations is higher than that of the chebyshev.

The same comparison has been made for a 10 element arrays with different spacing and the results has been shown in the table (6) below.

Table 6 : Comparison between uniform,binomial and chebyshev 10 element array at different spacing

Elements spacing (λ)	Uniform Array ref.3				Binomial Array ref.4				Chebyshev Array			
	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL
0.1	6.56	54	-17.5	2	2.72	0	-∞	0	5.89	60	-∞	0
0.2	9.48	24	-3.33	4	6.21	52	-∞	0	9.33	28	-10.8	4
0.3	11.21	18	-1.9	8	8.32	32	-∞	0	11.1	20	-8.75	8
0.4	12.43	12	-0.52	12	9.69	24	-∞	0	12.31	12	-7.52	12
0.5	13.36	8	0.29	16	10.7	20	-∞	0	13.22	12	-6.37	16
0.6	14.11	8	1.14	20	11.2	16	-∞	0	13.96	8	-5.93	20
0.7	14.71	8	1.74	24	12.2	16	-∞	0	14.58	8	-5.37	24
0.8	15.17	4	2.19	28	12.7	12	-3.17	2	15.12	8	-4.97	28
0.9	15.34	4	2.26	32	11.9	12	8.23	2	15.58	4	-0.15	34
1	11.43	4	-1.77	32	9.08	10	-∞	0	11.30	4	-8.45	28

At 1.8GHz operating frequency it is clear that the optimum separation for uniform and chebyshev excitations is 0.9λ which gives the best gain, but for binomial array it was 0.8λ

The same comparison has been made for different of element arrays with different spacing and the results has been shown in the table (7) below.

Table 7 : Comparison between uniform,binomial and chebyshev 10 element array at different spacing

No. of element	Uniform Array ref.3					Binomial Array ref.4					Chebyshev Arrat				
	Elements spacing (λ)	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL	Elements spacing (λ)	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL	Elements spacing (λ)	Gain (dBi)	HPBW (ver. plane)	SLL max. (dBi)	No. of SLL
2	0.66	7	44	0.73	2	0.66	7	44	0.73	2	0.66	7	44	0.73	2
3	0.75	9.35	24	0.11	6	0.72	8.81	30	1.15	2	0.72	9.20	28	0.23	6
4	0.8	10.9	16	-0.61	10	0.74	9.88	24	0.31	2	0.75	10.8	20	-3.1	10
5	0.82	12	12	-0.22	14	0.74	10.6	20	-2.07	2	0.8	12.1	16	-2.04	14
6	0.86	13	10	0.33	18	0.75	11.2	18	-3.31	2	0.84	13.1	12	0.26	18
7	0.88	13.8	8	0.87	22	0.78	11.7	16	-1.33	2	0.86	13.9	10	-0.12	22

8	0.9	14.4	8	2.93	26	0.78	12.1	14	-3.17	2	0.86	14.5	8	-5.63	26
9	0.9	14.9	6	1.82	30	0.8	12.4	12	-1.68	2	0.88	15.1	8	-5.12	30
10	0.92	15.4	4	4.19	34	0.82	12.8	12	0.13	2	0.9	15.6	6	-0.15	34

From the above table it is shown that the optimum separation for three method are difference for example in 10-element for chebyshev arrays was (d=0.9λ), which gave the best radiation properties while for binomial arrays the optimum space dimension was (d=0.8λ), uniform array with space between elements (d=0.92λ) had best radiation properties. The sequence of high gain and good HPBW starts from uniform excitation to chebyshev then binomial arrays.

IV. Conclusions

1. At 0.5λ spacing the sequence of high gain and good HPBW starts from uniform excitation to chebyshev then binomial arrays.
2. Binomial arrays have very low side lobes compared with chebyshev and uniform excitation arrays because the excitation coefficients in binomial arrays are very large especially in long arrays which cause great different in level between major lobe and side lobe in radiation properties. Moreover there is no SLL level for 10-element when space from (0.1λ until 0.7λ) after this point produce SLL, but in practical it's difficult to produce signals with great difference between the excitation coefficient. For 10-elements binomial array the different between center coefficient ($a_0 = 126$) and last excitation coefficient ($a_5 = 1$) is too much, while in chebyshev arrays for same number of elements, the center coefficient $a_0 = 1.5579$) and the last coefficient ($a_5 = 1$), in practical its easy generate signals with variation in amplitude between the center and the edge.
3. When the spacing between the elements was λ, all the three different excitations produce both broadside and end fire radiation patterns together.
4. For a fixed element spacing at 0.5λ the number of side lobes for both uniform and chebyshev excitations increase equally by two lobes per each number increment of elements but with different intensities such that the intensity of the side lobes for the uniform array excitations is higher than that of the chebyshev while the binomial array has not any side lobes.
5. When the spacing between the elements increases from 0.1λ to 0.8λ at fixed fix number of 10 element, both the uniform and chebyshev arrays have the same rate of increase of side lobes with the sequence of high gain from uniform excitation to chebyshev then binomial arrays.
6. For any number of elements up to 10 and the best spacing for high gain of each one both uniform and chebyshev arrays have almost the gain which was higher than that of the binomial array.
7. For the same upper condition both uniform and chebyshev arrays have the same rate of increase of side lobes which was four lopes per each number of elements while the binomial one has a fixed number of two side lobes.

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Hybrid Masking Algorithm for Universal Hearing Aid System

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Abstract- Technological advances witnessed in cochlear implant (CI) devices, most CI users can now achieve reliable speech intelligibility in controlled quiet scenarios, particularly in predictable conversations. Environmental distortions, such as reverberation and additive noise, on the other hand, are known to significantly degrade speech intelligibility. This paper validates a novel approach to predict speech intelligibility for Cochlear Implant users (CIs) in reverberant environments. First, energy thresholding is proposed to reduce the variability caused by the differences in modulation spectral representations for different phonemes and speakers, as well as speech enhancement algorithm artifacts. Second, a narrower range of Fourier domain based echo Filter is employed to reduce fundamental frequency effects. Results indicate substantial improvements in intelligibility over that attained by human listeners with unprocessed stimuli. The findings from this study show that algorithm can estimate reliably the SNR and can improve speech intelligibility.

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GJRE-F Classification : FOR Code: 090699

Strictly as per the compliance and regulations of :

Hybrid Masking Algorithm for Universal Hearing Aid System

H. Hensiba [α] & Mrs. V. P. Birla [σ]

Abstract- Technological advances witnessed in cochlear implant (CI) devices, most CI users can now achieve reliable speech intelligibility in controlled quiet scenarios, particularly in predictable conversations. Environmental distortions, such as reverberation and additive noise, on the other hand, are known to significantly degrade speech intelligibility. This paper validates a novel approach to predict speech intelligibility for Cochlear Implant users (CIs) in reverberant environments. First, energy thresholding is proposed to reduce the variability caused by the differences in modulation spectral representations for different phonemes and speakers, as well as speech enhancement algorithm artifacts. Second, a narrower range of Fourier domain based echo Filter is employed to reduce fundamental frequency effects. Results indicate substantial improvements in intelligibility over that attained by human listeners with unprocessed stimuli. The findings from this study show that algorithm can estimate reliably the SNR and can improve speech intelligibility.

Keywords: cochlear implants(CIs), noise reduction, speech enhancement, wiener filter(WF), SNR.

I. Introduction

UNDERSTANDING speech in adverse listening conditions with auditory prostheses like hearing aids or CIs is a very which try to remove as much noise as possible from the mixture of the target speech and the interfering sound, with the objective of increasing SI and/or improving the speech quality of the processed signal. A typical constraint of such strategies is that distortions of the target signal should be avoided.

Usually, noise reduction algorithms operate upon a time frequency representation of the input (noisy) signal, applying a gain to each time frequency point to suppress the noise. The pattern of the gain function over all time frequency points is often called mask. Most of these time frequency domain approaches derive their gains as a function of the short-term signal-to-noise ratio (SNR) in the respective time frequency point. There is an ongoing discussion about the choice of the perfect gain function that improves SI and speech quality in NH listeners very popular choice is the so-called binary mask (BM).

The mask is motivated by the auditory masking phenomenon and preserves with its binary values time frequency points, where the target is dominant (i.e., the short- term SNR is above a threshold). The BM exploits the spar sity and disjointness of the target and interferer spectra. When a priori knowledge of the signal and noise spectra is used in the derivation of the mask, the mask is often called IBM. Under certain listening conditions, approaches based on BMs with and without a priori knowledge for the mask computation can increase SI in NH, and hearing impaired listeners.

In contrast to the hard-decision approach of the BM, state-of- the-art noise reduction algorithms derive mask. Such algorithms demonstrate improved speech quality as compared to BM processed output. A popular representative of this class of algorithms is the Wiener filter (WF), which was shown to be very promising in terms of quality improvement. When a priori knowledge was used to calculate the gain function of the WF, the approach is referred to as IWF. It was shown in that the IWF restored perfect intelligibility with a Bark-scale frequency resolution even at very low SNRs in both multi talker babble noise and interfering talker scenarios. This was in stark contrast to the performance of the IBM, which yielded intelligibility scores of around 60% at the low SNRs. In this study, the potential of the IWF and the IBM approaches in terms of SI and speech quality is investigated with regard to its application in CIs.

The tests are carried out on two groups of participants: a group of NH subjects listening to noise vocoder simulations a same model of CI processing of the processed signals, and a group of CI users. Because of the relative ease with which NH volunteers may cruited, tests on NHlisteners, presented with noise vocoded versions of the processed signals comparable to CI processing, are often used in a first step to evaluate speech enhancement strategies for application in CIs. In our case, the inclusion of such tests with noise vocoded sounds (also referred to as CI simulations) allows us to investigate if the intelligibility scores obtained on NH listeners translate to the scores of CI users as well.

The aims of this study are the following: we wish to vestigate in terms of SI, which mask pattern is more beneficial for NH subjects listening to noise vocoder CI simulations and for CI users. It is interesting in particular for CI users, because noise reduction approaches based on time frequency masks can be added to the signal processing chain of existing clinical coding strategies without significant effect on other stages. Furthermore, we study the influence of estimation errors

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on the SI for both groups of listeners, as it was shown in that CI users are less sensitive to speech distortions. The design of the study allows us to investigate if the SI results ob tained with NH listeners using CI simulations can be translated to that of CI users. Additionally, we want to study the potential for speech quality improvement of both mask patterns in CI users.

II. Signal Processing

In Cochlear, Ltd., up to N=22 envelopes are extracted in the frequency range up to 8 kHz. Therefore, such CIs usually operate with a frequency resolution that is close to the Bark- scale spectrum used in. The signal model and processing used in this study are very similar to that in purposes of completeness, we present these briefly below.

a) *Signal Model*

Denote the time discrete signal recorded by the microphone as y(t), where t is the sample index. The signal y(t) consists of the target signal s(t) and the additive interference v(t). This additive signal model for the recorded signal can be written as

$$y(t)=s(t)+v(t). \quad (1)$$

Due to the fact that the IBM and IWF speech enhancement approaches operate in the frequency domain, the short time frequency representation of the signal in (1) can be written, with the frame index n and the frequency index k, as

$$Y(n,k)=S(n,k)+V(n,k). \quad (2)$$

Y (n,k) is the microphone signal in the time frequency domain, S (n,k) and V (n,k) represent the target signal and the interferer, respectively.

The estimate ˆ S(n,k) of the target signal is obtained by applying the time frequency mask G(n,k) ∊ [0,1] yielded by the IBM and/or IWF approach, to Y (n,k). Thus, the output of the speech enhancement step can be written as ˆ

$$S(n,k)=G(n,k)Y(n,k). \quad (3)$$

Both the IBM and the IWF approaches derive their respective masks as a function of the short-term SNR ξ (n,k), which is defined as the ratio between the power spectral density (PSD) of the target signal ΦSS (n,k) and the PSD of the interferer ΦVV (n,k)

$$\xi(n,k)=\Phi SS(n,k)\ \Phi VV(n,k). \quad (4)$$

Usually, the PSD of the target signal and the interfering sound are computed by using the Welch method, one implementation which is a first order recursive smoothing of the respective periodograms. Since we deal with ideal estimates of the parameters of the IBM and the IWF approach, we can approximate the PSD with

$$\Phi SS(n,k)=|S(n,k)|2 \quad (5)$$

$$\Phi VV(n,k)=|V(n,k)|2. \quad (6)$$

i. *Ideal Binary Mask*

The IBM GIBM consists of binary weights. GIBM is equal to 1 when the SNR is above a threshold value, and 0 when the SNR is lower than this threshold. In this study, the threshold used was the global input SNR ξin. The BM GIBM can be written as

$$GIBM(n,k) = 1, \text{if } \xi(n,k) \geq \xi \text{ in } 0, \text{else}. \quad (7)$$

Note that for a given combination of s(t) and v(t), the mask pattern is constant and independent of the SNR. This is termed as the local threshold. The binary gain function GIBM is applied to the input signal Y to obtain the enhanced output ˆ SIBM ˆ

$$SIBM(n,k)=GIBM(n,k)Y(n,k). \quad (8)$$

ii. *Ideal Wiener Filter*

The gain function GIWF of the WF approach is a continuous value between 0 and 1. It is obtained as the minimum mean-squared error estimate of the complex spectral amplitude

$$\min E\{|S(n,k)-\hat{\ } S(n,k)|2\} \quad (9)$$

and can be written as

$$GIWF(n,k) = \xi(n,k)\ 1+\xi(n,k) \quad (10)$$

The corresponding estimate ˆ SIWF may then be written as ˆ

$$SIWF(n,k)\ GIWF(n,k)Y(n,k) \quad (11)$$

iii. *Simulation of Estimation Errors*

To investigate the influence of estimation errors in the mask patterns of WF and the BM on SI, such errors in the mask pattern were simulated. Due to the fact that over and under estimation errors influence SI differently we use the approach first described is to generate a balanced pattern of estimation errors. For the mask derivation, the spectra of the target and the noise signal were corrupted with an additional noise term which can be written as ˜

$$S(n,k)=S(n,k)+S(k) \quad (12)$$

$$V(n,k) = V(n,k)+V(k) \quad (13)$$

where S (k) and V (k) are complex randomly distributed variables with zero mean and power equal to the respective clean signal in the frequency band k. The corrupted spectra influence the short-term SNR estimation in and, thereby, the mask computation. This results in corrupted mask patterns GBM and GWF. When referring to results and patterns obtained in the condition with perturbed estimates, the masks are called BM and WF. The output signals for the BM and the WF mask are ˆ

$$SBM\ (n,k) = GBM\ (n,k)Y\ (n,k) \quad (14)$$

$$SWF\ (n,k) = GWF\ (n,k)Y\ (n,k) \quad (15)$$

The corrupted parameter estimates are only used for the mask pattern estimates. The corrupted masks are applied to the original, unperturbed mixture in (14) and (15). This manner of simulating estimation errors allows for both under and over estimation of the instantaneous PSD estimate. Additionally, such a perturbation of the underlying spectrum has the advantage that it does not preserve the silence periods of the speech and/or interference. Thus, the musical-noise phenomenon will be present in such speech/interference pauses [26]. This lends realism to the simulation.

b) General Processing Steps

The processing steps to generate the stimuli that are presented to the listener are shown in Fig. 1. The first six processing steps are the same for both groups of listeners. The processing steps that are different between NH listeners and CI users are represented by the black trace for the noise band vocoder CI simulation with NH listeners and by the dashed gray trace for the electrical stimulation with CI users.

In the first step, the target signals(t) and the interfering signal v(t) (sampled at 16 kHz) are filtered with a pre emphasis filter that consists of the frequency response of the SP12 microphone of the Freedom speech processor of Cochlear, Ltd. The result of this pre emphasis is a boost of the higher frequencies. A square-root Hann window was used as the analysis window. The envelope extraction is done by grouping the magnitude-squared DFT coefficients into N frequency bands. This process is applied to the target and the interfering signal to calculate the power spectral density (PSD) estimates used for the gain computation in (7) and (10).

c) Noise Band Vocoder as CI Simulation

The number of channels used for the noise band vocoder CI simulation was set to N =8, because asymptotic SI performance for most CI users is reached with the current clinical speech processing strategies with eight effective channels [27]. The cut off frequencies to obtain the band pass filtered envelopes are 187.5, 437.5 7937.5 Hz. These cutoff values for the band pass filters correspond to bandwidths of 250, 250, 375, 500, 750, 1125, 1750, and 2750 Hz for the eight channels. The signal components under 187.5 Hzare not considered in the signal processing. Finally, all noise vocoded channels are added to obtain the final audio stimulus that can be presented acoustically to the NH listener.

d) Cochlear Implants

The current clinical CI device of Cochlear, Ltd., can stimulate 22 channels. Therefore for most patients, N = 22frequency bands are processed in the envelope extraction stage. In this study, all six patients used a frequency resolution of 22 channels. The advanced combination encoder (ACE) strategy that is the default speech processing strategy in CIs of Cochlear, Ltd., does not stimulate all available channels in each time frame. The ACE strategy consists of a maxima selection stage in each frame, where the M<N channels with envelopes of the highest amplitude in the respective frame are selected and only these channels are stimulated. In clinical practice, the number M of selected maxima varies between 7 and 12. In this study, all patients used M =8 design must contain these enable conditions in order to use and benefit from clock gating. This clock gating process can also save significant die area as well as power, since it removes large numbers o replaces them with clock gating logic. This clock gating logic is generally in the form of "Integrated clock gating" (ICG) cells. However, note that the clock gating logic will change the clock tree s

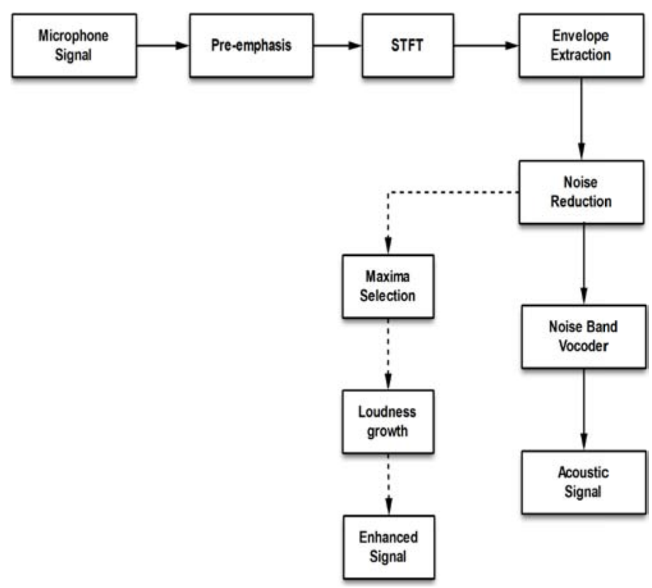


Fig. 1 : Processing chain for experiments with a noise band vocoder in the top line for NH listeners (black trace) and the bottom line for CI listeners (dashed gray trace)

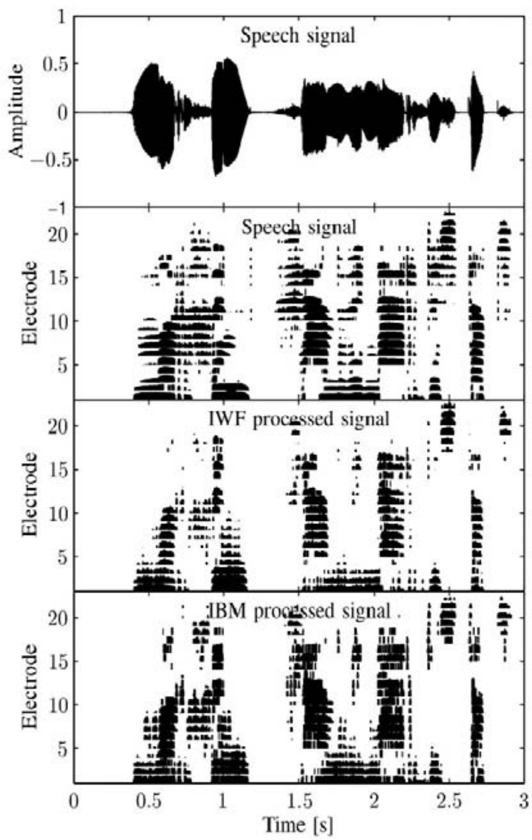


Fig 2 : Example of the speech signal

III. Methods

a) Subjects

The sentence recognition task with the noise vocoded CI simulation was conducted with six NH listeners. All NH subjects were between 16 and 22 years old (mean age 20.1 years) and had hearing thresholds below 20 dB hearing level (HL), on both ears, for the octave frequencies between 125 and8000 Hz. They were not paid for their travel expenses. The second group of listeners consisted of six Clusers. The group of CI users was paid for their participation. All subjects were Dutch speaking adults. They signed an informed consent form before the tests were conducted.

b) Test Material

The Leuven Intelligibility Sentence Test (LIST) sentences were selected as the target speech material. These are Dutch/Flemish sentences spoken by a male and a female speaker. The female LIST sentence material consists of 35 lists of ten Dutch/Flemish sentences. There are 38 lists of ten Dutch/Flemish sentences available in the male LIST sentence material of which the vocabulary of 20 lists is different from the female lists. Only these non-overlapping lists were used during the experiments. Each sentence for the male and the female speaker consists of four to eight words. Keywords are marked in each sentence which results in 32 to 33 keywords per list.

Each list is balanced according to the phonetic distribution of conversational speech. The scoring was done on sentence level, where a sentence was marked as correctly recognized if all keywords of the respective sentence were repeated correctly. In the speech recognition task in multi talker babble noise, the female LIST sentences were used as the target speech material and the Auditec multi talker babble noise (from the CD Auditory Tests (Revised), Auditec, St. Louis, MO,USA) was used as the interfering sound. In the speech in speech scenario, the male LIST sentences served as the target speech material, while the female lists were the interfering speaker. The lists that did not serve before as the target speech material in the sentence recognition task in multi talker babble noise were used as the interferer.

c) Procedure

The SI improvement was evaluated in two sentence recognition tasks: a sentence recognition task when the mask patterns were derived with ideal parameter estimates, and a sentence recognition task with simulated estimation errors in the mask calculation. Both groups of listeners participated in these tasks. A quality rating was performed to as the potential for speech quality improvement by the IWF and the IBM with CI users. All listening tasks were conducted in a test-retest design of the study with a break of at least one week between the two sessions, each session lasting around 90 min. In all listening tasks, the processed signals for all algorithms were rescaled to the same presentation level of the clean target speech. This was done to prevent audibility issues of the pro- cessed signals at low SNRs. The presentation level was set to 65 dB sound pressure level. After the processing steps described previously, the signals were presented monaurally (left ear) using Sennheiser HDA200 headphones, in a sound-proof booth, for the NH participants. With CI users, direct stimulation of the channels of the CI was done with the L34 research processor.

i. *Speech Intelligibility*

In this first sentence recognition task, the signals were mixed at SNRs of 0 dBto−20 dB or −25 dB in steps of 5 dB for the noice vocoded speech and the CI users, respectively. The mask patterns for the IWF and IBM were constructed with ideal parameter knowledge. After the processing steps described previously, the signals were presented monaurally (left ear) using Sennheiser HDA200 headphones, in a sound-proof booth, for the NH participants.

ii. *Robustness to Estimation Errors:*

In this second task, both groups of listeners were tested at SNRs of 0 and−5 dB for the WF and the BM processed signals. The interfering signal was multi talker babble noise. The SNRs chosen in this study represent more realistic listening environments as compared to the first task.

iv. *Preference Rating*

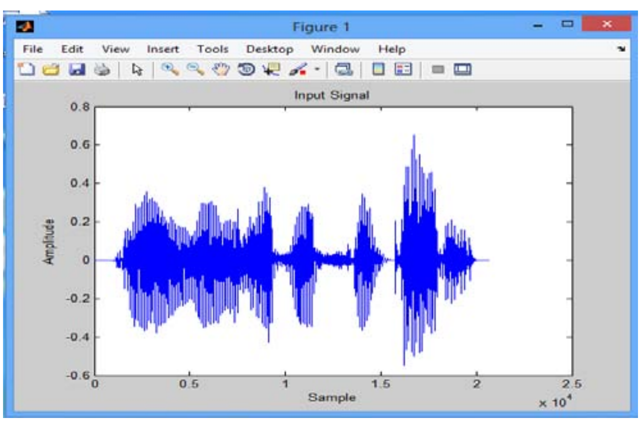
We also investigated the speech quality of the IWF and the IBM approaches. For this, a preference rating was done only with the CI users. Preference is often used as a model of quality .Therefore it is assumed that a preference in a pair wise comparison correlates with a quality advantage of the preferred signal. For the quality improvement, the same procedure was used. This consists of a two-stage pair wise preference rating test. The pair wise comparison was administered across

1) clean speech and IWF processed output;
2) clean speech and IBM processed output;
3) IWF processed output and IBM processed output.

Clean speech, as implied here, corresponds to the clean tar- get sentence without any interfering signal present and without being subject to any noise reduction. Two noise conditions were tested (speech-in-speech, speech-in-babble) at an SNR of 0 dB . First, the signals of the respective pairwise comparison were played one after the other and the subject had to make a decision on which one was preferred in terms of quality.

IV. Results

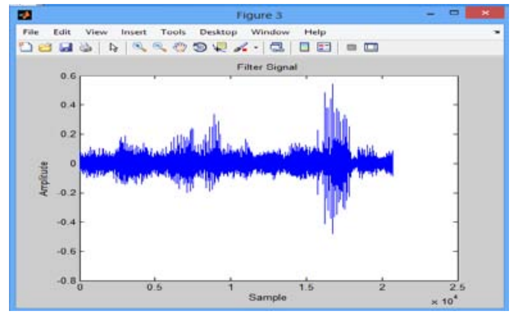
a) Input signal



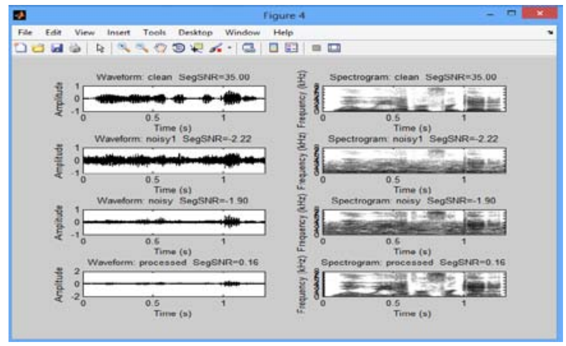
The input signal to select the normal speech in audio signal. It is using to normal speech processing at the signal module compaired to audio speech.

b) Filter signal

Add the babble noise and filter to the normal speech. In non hearing person get the normal speech signal to add the sum noise and clearing to the audio. And filtering to the noise to send the clear speech in CI users.

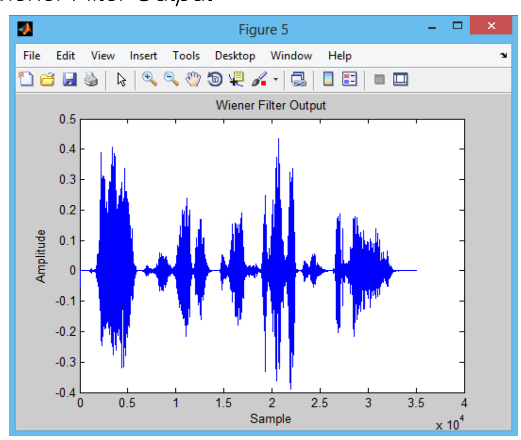


c) Speech output

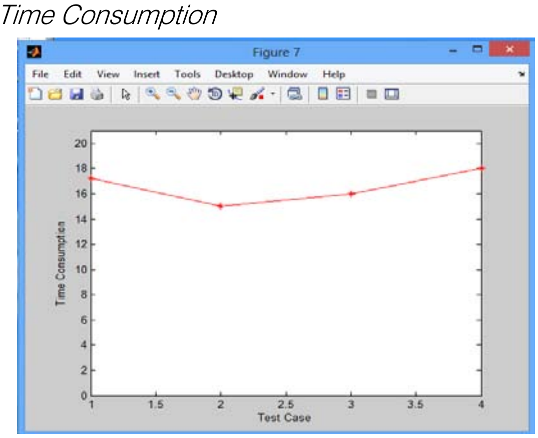


Speech output that compared to Ideal Binary Mask and Ideal Wiener Filter. IBM compared to binary weights in amplitude value. IWF compared to continuous weights in frequency value. Multi talker babble was used as the interference.

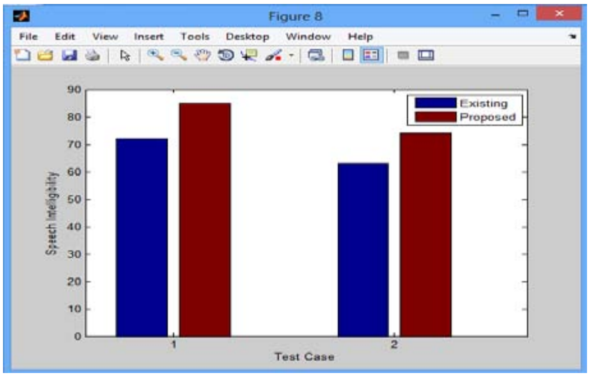
d) Wiener Filter Output



e) Time Consumption



f) Speech Intelligibility



V. Conclusion

The investigated the potential of the IWF and the BM approaches for SI and speech quality improvement in CIs. The results of NH listeners presented with noise vocoded CI simulations are consistent and favor the soft decision approach over the BM. However, the outcomes of this study suggest that for CI users with the ACE channel selection strategy the choice between a hard and a soft decision approach is not important in terms of SI. This study also points out that the frequency resolution of the noise reduction algorithm is a less important parameter in the application of CI than it seems to be in NH listeners. The obtained results are a reference for SI evaluations of time frequency masking algorithms applied to noisy signals without a priori knowledge.

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Detecting and Recognizing the Face and Iris Features from a Video Sequence using DBPNN and Adaptive Hamming Distance

By S. Revathy

Abstract- Dense feature extraction is becoming increasingly popular in face recognition. Face recognition is a vital component for authorization and security. In earlier days, CCA (Canonical Correlation Analysis) and SIFT (Scale Invariant Feature Transforms) was used for face recognition. Since multi scale extraction is not possible with these existing methods, a new approach to dense feature extraction is developed in this project. The proposed method combines dense feature extraction and decision based propagation neural network (DBPNN). Neural network algorithm is presented to recognize the face at different angle, and it is used for training and learning and leading to efficient and robust face recognition. Finally Iris matching is done by using Iterative randomized Hough transform for detecting the pupil region with number of iteration counts. Experimental results show that the proposed method is providing effective recognition rate with accuracy in comparing with existing methods.

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GJRE-F Classification : FOR Code: 290903

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I. Introduction

The human face plays an important role in our social interaction for conveying people's identity. Face recognition is an ability to recognize people by their unique facial characteristics. Biometric-based technologies include identification based on physiological characteristics such as face, fingerprints, finger geometry, hand geometry, hand veins, palm, iris, retina, ear and voice and behavioural traits such as gait, signature and keystroke dynamics. Face recognition can be done passively without any explicit action or participation on the part of the user since face images can be acquired from a distance by a camera. Iris and retina identification require expensive equipment and are much too sensitive to any body motion. Voice recognition is susceptible to background noises in public places and auditory fluctuations on a phone line or tape recording. Signatures can be modified or forged. However, facial images can be easily obtained with a couple of inexpensive fixed cameras. Face recognition is totally non-intrusive and does not carry any such health risks.

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II. Related Works

Lowe used a modification of the k-d tree algorithm in this method and is called the Best-bin first search method that can identify the nearest neighbors with high probability using only a limited amount of computation [12]. The BBF algorithm uses a modified search ordering for the k-d tree algorithm so that bins in feature space are searched in the order of their closest distance from the query location. This search order requires the use of a heap-based priority queue for efficient determination of the search order. The best candidate match for each key point is found by identifying its nearest neighbour in the database of key points from training images.

When an image contains complex background, the SIFT descriptors tend to spread over the entire image rather than being concentrated in the object region. As a result, the actual object can be neglected in the matching process. Since the number of extracted SIFT descriptors is typically large, the computational cost to match extracted key-points is very high.

Adaptive matching algorithm is used to match the person according to the trained values. The test image feature is compare with the trained image dataset if the test image feature and trained image feature is similar the person is matched person else the person is not matched. trained When the probe image is given, the algorithm may compare the features of the input image with the trained database and determines the results as authenticated [18]. The algorithm may conclude its results only after all the features in the image are matched and hence this process is much superior to others. Different poses are not applicable in this method.

The template matching is simple technique for image processing topics like feature extraction, edge detection, object extraction. Template matching can be subdivided between two approaches: feature-based and template-based matching. The feature-based approach uses the features of the search and to recognize a human face, some special features need to be extracted. These special features include eyes, nose, mouth, and chin along with the shape of the face.

Initially, the subject image is enhanced and segmented. Then the contour features of the face are extracted by contour extraction method, and compares with the extracted features of the database image [4]. If there is a match then the person in the subject image is recognized.

III. Proposed Method

a) Face Recognition from Video

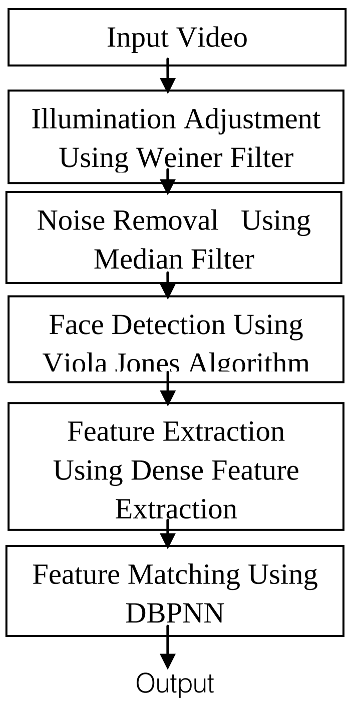

Figure 1 : Block diagram for face recognition

i. *ILLumination Adjustment*

An important goal of this adjustment is to render specific colours particularly neutral colours correctly; hence, the general method is sometimes called gray balance, neutral balance, or white balance.

a. *Wiener Filter*

The most important technique for removal of blur in images due to linear motion is the Wiener filter. The first process that was performed was creating a point spread function to add blur to a video. The blur was implemented by first creating a PSF filter in Mat Lab that would approximate linear motion blur. This PSF was then convolved with the original image to produce the blurred video. Convolution is a mathematical process by which a signal, in this case the image is acted on by a system the filter in order to find the resulting signal. The amount of blur added to the original video depends on two parameters of the PSF are the length of blur in pixels, and the angle of the blur. After a known amount of blur was introduced into the video, an attempt was made to restore the new blurred video to its original form. The Weiner filter is an inverse filter that employs a linear deconvolution method. Linear deconvolution means that the output is a linear combination of t two common types of impulse noise are the salt-and-pepper noise and the random-valued noise. For images corrupted by salt-and-pepper noise respectively random-valued noise, the noisy pixels can take only the maximum and the minimum values respectively any random value in the dynamic range. There are many works on the restoration of images corrupted by impulse noise. See, for instance, the nonlinear digital filters.

ii. *Noise Removal Using Median Filter*

The median filter was once the most popular nonlinear filter for removing impulse noise, because of its good denoising power and computational efficiency. However, when the noise level is over 50%, some details and edges of the original image are smeared by the filter. Different remedies of the median filter have been proposed, like the adaptive median filter the multi-state median filter or the median filter based on homogeneity information. These so-called decision-based or switching filters first identify possible noisy pixels and then replace them by using the median filter or its variants, while leaving all other pixels unchanged. These filters are good at detecting noise even at a high noise level. Their main drawback is that the noisy pixels are replaced by some median value in their vicinity without taking into account local features such as the possible presence of edges. Hence details and edges are not recovered satisfactorily, especially when the noise level is high.

For images corrupted by Gaussian noise, least-squares methods based on edge-preserving regularization functional have been used successfully to preserve the edges and the details in the images. These methods fail in the presence of impulse noise because the noise is heavy tailed. Moreover the restoration will alter basically all pixels in the image, including those that are not corrupted by the impulse noise. Recently, non-smooth data-fidelity terms have been used along with edge preserving regularization to deal with impulse noise.

We propose a powerful two-stage scheme which combines the variation method proposed in with the adaptive median filter. More precisely, the noise candidates are first identified by the adaptive median filter, and then these noise candidates are selectively restored using an objective function with a data fidelity term and an edge-preserving regularization term. Since the edges are preserved for the noise candidates, and no changes are made to the other pixels, the performance of our combined approach is much better than that of either one of the methods. Salt and pepper noise with noise ratio as high as 90% can be cleaned quite efficiently.

iii. *Face Detection*

Face-detection algorithms focus on the detection of frontal human faces. It is analogous to image detection in which the image of a person is matched bit by bit. Any facial feature changes in the database will invalidate the matching process.

a. *Viola-Jones Algorithm*

The viola-Jones object detection framework is the first object detection framework to provide competitive object detection rates in real-time primarily by the problem of face detection. The human can do this easily, but a computer needs precise instructions and constraints. To make the task more manageable, viola-Jones requires full view frontal upright faces. The algorithm has four stages, Haar Feature Selection, Creating an Integral Image, Adaboost Training, and Cascading Classifiers. The features of the detection framework involve the sums of image pixels within rectangular areas and it bear some resemblance to Haar basis functions, which have been used previously in the realm of image-based object detection. However, since the features used by Viola and Jones all rely on more than one rectangular area, they are generally more complex. The value of any given feature is always simply the sum of the pixels within clear rectangles subtracted from the sum of the pixels within shaded rectangles. The rectangular features of this sort are rather primitive when compared to alternatives such as steerable filters. Although they are sensitive to vertical and horizontal features, their feedback is considerably coarser.

The Haar features are the features which are used to match the similar properties of the human faces. Some of the similar properties related to human face are generally considered as, the eye region is darker than the upper-cheeks and the nose bridge region is brighter than the eyes. Composition of properties forming matchable facial features is, Location and size are eyes, mouth, and bridge of nose Value is oriented gradients of pixel intensities.

Rectangle features considered is given by the equation Value = Σ (pixels in black area) - Σ (pixels in white area).Three types: two-, three-, four-rectangles, Viola & Jones used two-rectangle features. For example: the difference in brightness between the white &black rectangles over a specific area where each feature is related to a special location in the sub-window.

An image representation called the integral image evaluates rectangular features in constant time, which gives them a considerable speed advantage over more sophisticated alternative features. Because each feature's rectangular area is always adjacent to at least one other rectangle, it follows that any two-rectangle feature can be computed in six array references, any three-rectangle feature in eight, and any four-rectangle feature in ten.

b. *Learning Algorithm*

The speed with which features may be evaluated does not adequately compensate for their number, however. For example, in a standard 24x24 pixel sub-window, there is a total of M=162,336 possible features and it would be prohibitively expensive to evaluate them all when testing an image. Thus, the object detection framework employs a variant of the learning algorithm Ad boost to both select the best features and to train classifiers that use them. This algorithm constructs a "strong" classifier as a linear combination of weighted simple "weak" classifiers.

$$h(x) = sin\left(\sum_{j=1}^{M} \propto j\; hj(x)\right) \qquad (1)$$

Each weak classifier is a threshold function based on the feature f_i

$$hj(x) = \begin{cases} -sj \text{ if } fj < \theta j \\ sj \text{ otherwise} \end{cases} \qquad (2)$$

The threshold value θj and the polarity $sj \in \pm 1$ are determined in the training, as well as the coefficients$\propto j$.

Here a simplified version of the learning algorithm is reported as,

Input: Set of N positive and negative training images with their labels

(x^i, y^i). If image i is a face$y^i = 1$, if not$y^i = -1$.

Initialization: assign a weight $w1^i = \frac{1}{N}$ to each image i.

For each feature *Fj* with J = 1,……, M

Apply the feature to each image in the training set, and then find the optimal threshold and polarityθ_j ,s_j that minimizes the weighted classification error. That is

$$\theta j, sj = \arg\min \sum_{i=1}^{N} wj^i cj^i \qquad (3)$$

Where,

$$cj^i = \begin{cases} 0 \text{ if } y^i = hj(x^i, \theta j, sj) \\ 1 \text{ otherwise} \end{cases}$$

Assign a weight $\propto_j$ to h_j that is inversely proportional to the error rate. In this way best classifiers are considered more.

The weights for the next iteration, i.e. w_{j+1}^i, are reduced for the images i that were correctly classified.

$$h(x) = sin\left(\sum_{j=1}^{M} \propto j\; hj(x)\right) \qquad (4)$$

iv. *Local Feature Extraction*

A feature is defined as an interesting part of an image, and features are used as a starting point for many computer vision algorithms. Since features are used as the starting point and main primitives for subsequent algorithms, the overall algorithm will often only be as good as its feature detector.

a. *Dense Feature Extraction*

Dense feature detection refers to a methods that aim at computing abstractions of image information

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and making local decisions at every image point whether there is an image feature of a given type at that point or not. The resulting features will be subsets of the image domain often in the form of isolated points, continuous curves or connected regions a feature is defined as an interesting part of an image and features are used as a starting point for many computer vision algorithms. Since features are used as the starting point and main primitives for subsequent algorithms, the overall algorithm will often only be as good as its feature detector. Consequently, the desirable property for a feature detector is repeatability whether or not the same feature will be detected in two or more different images of the same scene.

Scale-space theory is a framework for multi scale signal representation developed by the Computer vision, image processing and signal processing communities with complementary motivations from physics and biological vision. It is a formal theory for handling image structures at different scales, by representing an image as a one-parameter family of smoothed images, the scale-space representation, parameterized by the size of the smoothing kernel used for suppressing fine-scale structures.

The parameter t in this family is referred to as the scale parameter, with the interpretation that image structures of spatial size smaller than about $\sqrt{t}$ have largely been smoothed away in the scale-space level at scale t.

v. *Decision Based Probablistic Neural Network*

A Probabilistic Neural Network is defined as an implementation of statistical algorithm called Kernel discriminate analysis in which the operations are organized into multilayered feed forward network with four layers: input layer, pattern layer, summation layer and output layer. A PNN is predominantly a classifier since it can map any input pattern to a number of classifications. Among the main advantages that discriminate PNN is Fast training process, an inherently parallel structure, guaranteed to converge to an optimal classifier as the size of the representative training set increases and training samples can be added or removed without extensive retraining.

Pattern layer/unit: there is one pattern node for each training example. Each pattern node/unit forms a product of the input pattern vector x (for classification) with a weight vector Wi, $Zi = x.Wi$, and when perform a non linear operation on Zi before outputting its activation level to the summation node/unit. Here instead of the sigmoid activation function of the nonlinear operation is used,

$$\exp\left[(Zi-1)/\sigma 2\right] \quad (5)$$

Both x and Wi are normalized to unit length, this is equivalent to using the

$$\exp\left[-(Wi-x)T(Wi-x)//2\sigma 2\right). \quad (6)$$

Summation layer/unit: each summation node/unit receives the outputs from pattern nodes associated with a given class. It simply sums the inputs from the pattern units that correspond to the category, from which the training pattern was selected,

$$\sum i, \exp\left[-\frac{(Wi-x)T(Wi-x)}{2\sigma 2}\right]. \quad (7)$$

Output layer/unit: the output nodes/units are two-input neurons. These units product binary outputs, related to two different categories $\Omega r, \Omega s, r \neq s, r, s = 1,2 \ldots . q$, by using the classification criterion,

$$\sum i, \exp\left[-\frac{(Wi-x)T(Wi-x)}{2\sigma 2}\right] > \sum j \exp\left[-\frac{(Wj-x)T(Wj-x)}{2\sigma 2}\right] \quad (8)$$

These units have only a single corresponding weight C, given by the loss parameters the prior probabilities and the number of training pattern in each category. Concretely, the corresponding weight is the ratio on a priori probabilities, divided by the ratio of samples and multiplied by the ratio of losses. This ratio can be determined only from the significance of the decision. There were developed non parametric techniques for estimating univariate probability density function from random samples. By using the multivariate Gaussian approximation, that is a sum of multivariate Gaussian distributions centered at each training sample, we obtain the following form of the pdf and i is the vector pattern number, m is the total number of training patterns, xi is the i^{th} training pattern from category class, $\Omega r, p$ is the input space dimension and σ is an adjustable smoothing parameter using the training procedure. The network is trained by setting the Wi weight vector in one of the pattern units equal to each x pattern in the training set and then connecting the pattern unit's output to the appropriate summation. In decision based PNN, if else condition is applied for the classification process where the classification can be done by the else condition format and the authentication is also based on the if else condition with more accuracy.

a. *Algorithm For Dbpnn*

Owned by one of the training sample and the weight vector initialize the objective function randomly generated contemporary parameter vector $Xi(i = 1,2 \ldots \ldots N)$and xi normalized samples training DBPNN, for videos, the objective function value for comparison with the objective function value of the current local optimal pests profile pests is, the objective function value is larger each particle, its objective function value of the objective function value of the current global optimum best compare update best objective function value larger

According to the formula updates the values of Xi *and* xi.

Termination of discrimination: If the number of iterations is the maximum number of times or the objective function reaches optimal if the loop is terminated, otherwise it returns Step3.

IV. Iris Recognition from Video

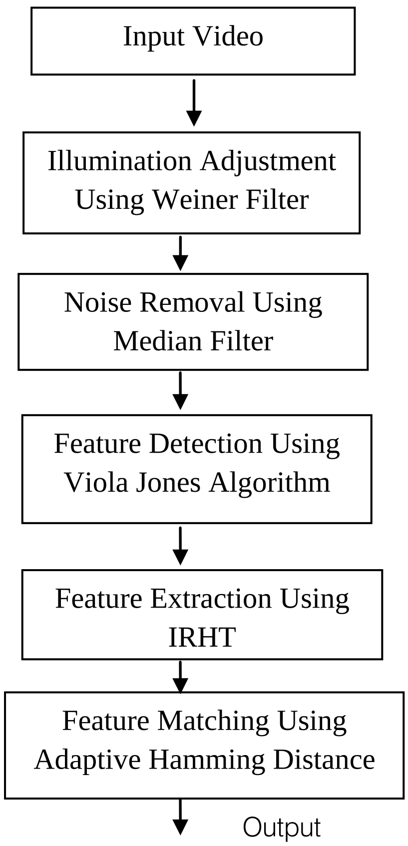

Figure 2 : Block diagram for iris recognition

a) Feature Extraction Using IRHT

An iterative randomized Hough transform (IRHT) is developed for detection of incomplete ellipses in images with strong noise. The IRHT iteratively applies the randomized Hough transform (RHT) to a region of interest in the image space. The region of interest is determined from the latest estimation of ellipse parameters. The IRHT zooms in on the target curve by iterative parameter adjustments and reciprocating use of the image and parameter spaces. During the iteration process, noise pixels are gradually excluded from the region of interest, and the estimation becomes progressively close to the target. The IRHT retains the advantages of RHT of high parameter resolution, computational simplicity and small storage while overcoming the noise susceptibility of RHT. Indivisible, multiple instances of ellipse can be sequentially detected. The IRHT was first tested for ellipse detection with synthesized images. It was then applied to fetal head detection in medical ultrasound images. The results demonstrate that the IRHT is a robust and efficient ellipse detection method for real-world applications.

Algorithm Steps

Iteration count

Mapping of edge points to the Hough space and storage in an accumulator.

Interpretation of the accumulator to yield lines of infinite length. The interpretation is done by thresholding and possibly other constraints.

Conversion of infinite lines to finite lines.

b) Feature Matching using Adaptive Hamming Distance

The weighted Hamming distance has been used for image retrieve, including Hamming distance weighting and the Anno Search. In each bit of the binary code is assigned with a bit-level weight, while the aim is to weight the overall Hamming distance of local features for image matching.

Only a single set of weights is used to measure either the importance of each bit in Hamming space, or to rescale the Hamming distance for better image matching. Jiang et al. propose a query-adaptive Hamming distance for image retrieve which assigns dynamic weights to hash bits, such that each bit is treated differently and dynamically. They harness a set of semantic concept classes that cover most semantic elements of image content. Then, different weights for each of the classes are learned with a supervised learning algorithm.

To compute the bit-level weights for a given query, a *k* nearest neighbour search is performed based on the original Hamming distance first, and then a linear combination of the weights of classes contained in the result list is used as the query-adaptive weights. In this section, we present the weighted Hamming distance ranking algorithm.

In most binary hashing algorithms, the distance between two points is simply measured by the Hamming distance between their binary codes. This distance metric is somewhat ambiguous, since for a K-bits binary code $H(p)$, there are $K\ m$ different binary codes sharing the same distance m to $H(p)$. In most binary hashing algorithms, each hash bit takes the same weight and makes the same contribution to distance calculation. On the contrary, in our algorithm, we give different bits different weights.

With the bit-level weights, the returned binary codes can be ranked by the weighted Hamming distance at In this section, we present the weighted Hamming distance ranking algorithm. In most binary hashing algorithms, the distance between two points is simply measured by the Hamming distance between their binary codes. This distance metric is somewhat ambiguous, since for a K-bits binary code $H(p)$, Km

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there are K different binary codes sharing the same distance m to $H(p)$. In most binary hashing algorithms, each hash bit takes the same weight and makes the same contribution to distance calculation.

On the contrary, in our algorithm, we give different bits for different weights. With the bit-level weights, the returned binary codes can be ranked by the weighted Hamming distance at 1585 1587 a finer-grained binary code level rather than at the original integer Hamming distance level. The bit-level weight associated with hash bit K is denoted as ωK. In the following, we will show that an effective bit-level weight is not only data dependent, but also query-dependent.

Note that, our algorithm is not to propose a new binary hashing method, but to give a ranking algorithm to improve the search accuracy of most existing binary hashing methods.

Some notations are given below to facilitate our discussion. Given a dataset $X = \{x(i)\}N, i = 1 \in$, the neighbor set of x is denoted as $N(x)$.

The paradigm of binary hashing is to first use a set of linear or non-linear hash functions $F = \{fk: Rd \to R\}Kk = 1$ to map$x \in X \; to \; F(x) \in RK$, and then binarize,

$F(x) = (f_{1(x)} \dots \dots f_{k(x)})T$ By comparing each with $f_{k(x)}$ a threshold Tk to get a K-bit binary code $H(x) \in \{0,1\}K$. Hence, the binary hash function is given by the, $h_{k(x)} = sgn(f_{k(x)} - T_k)$ we call the$f_{k(x)}$ unbinarized hash value. Each dimension of $H(x)$ is called a hash bit, and for a query q and its neighbour p, if the k-th bit of $H(q)$ and $H(p)$ is different, we call there is a bit-flipping on hash bit k. The weighted Hamming distance between two binary codes $h(1)$and $h(2)$is denoted as $D^w H(h(1), h(2))$.

V. Expreimental Results

a) Simulation Results of Face Recognition

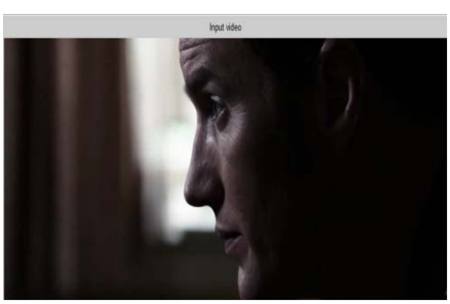

Figure 3 : Input video for face

Figure 4 : Deblurred face

The above figure 3 shows the input video taken for face recognition from side pose and the figure 4. Shows the deblurred face and here the blur is implemented by first creating a PSF filter in Mat Lab that would approximate linear motion blur.

This PSF was then convolved with the original image to produce the blurred image the most important technique for removal of blur in images due to linear motion is the Wiener filter and this process is called the illumination adjustment.

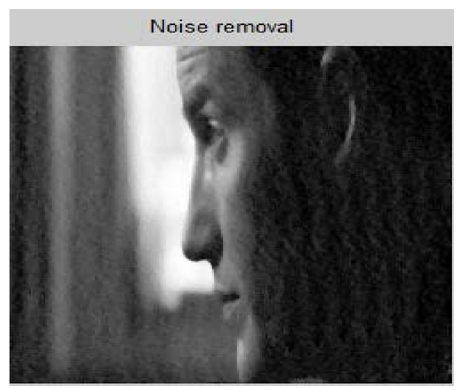

Figure 5 : Removal of noise from face

Figure 6 : Face detection

The figure 5 shows the removal of salt and pepper noise from face image using median filter and the figure 6. shows the image of face detection from video using viola-Jones algorithm.

The figure 7. shows extraction of local features from detected face using multi scale feature extraction method the features from chin, cheek and ear points are considered and in the figure. 8 the authentication of face image is done by using DBPNN method.

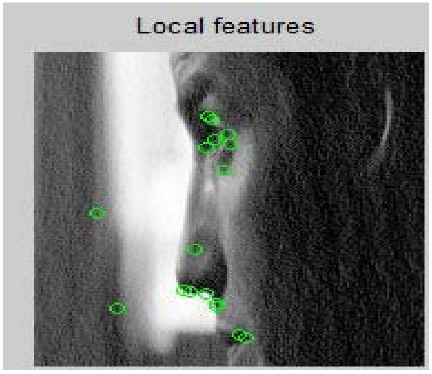

Figure 7 : Extraction of local features

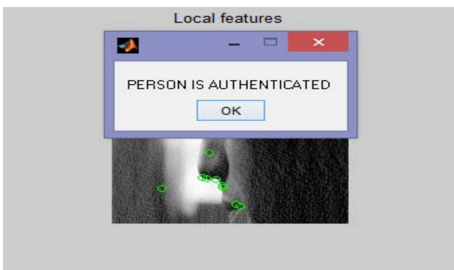

Figure 8 : Face authentication

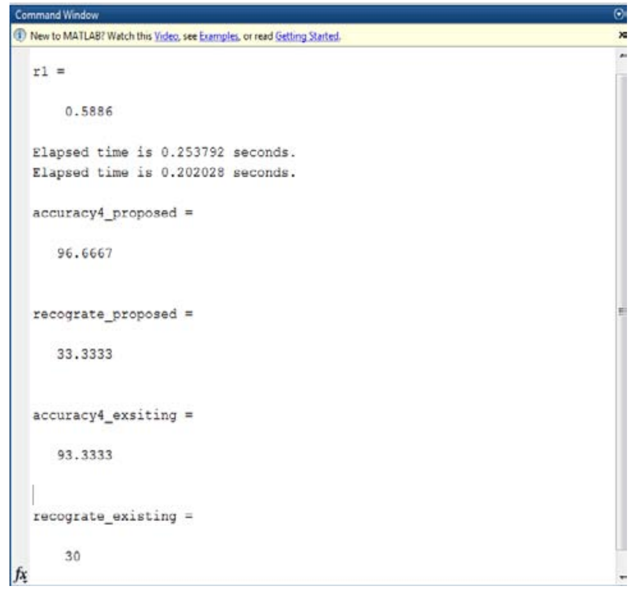

Figure 9 : Command window from mat lab

The above figure shows the output command window obtained by running the program coding and these values are used for evaluating the performance of the proposed methods applied.

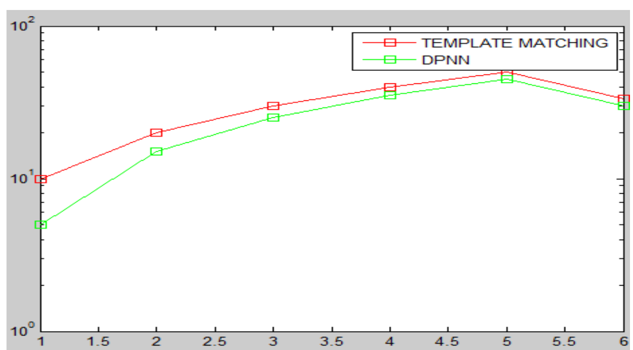

Figure 10 : performance graph of template matching versus DBPNN

Figure.10 gives the performance evaluation of template matching with DBPNN where the graph is plotted between accuracy and recognition rate.

b) Simulation Results of Iris Recognition

The figure.11 is the input video considered for iris recognition process and the figure.12 is the deblurred image from the liner motion blurred image using wiener filter.

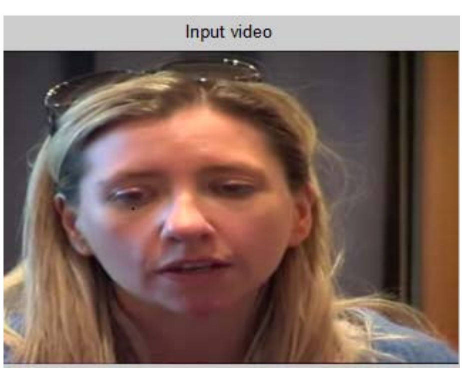

Figure 11 : Input video for iris

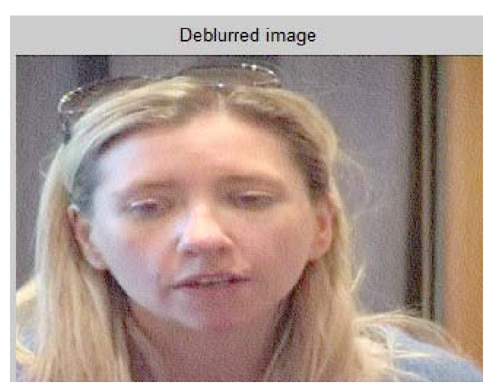

Figure 12 : Deblurred images

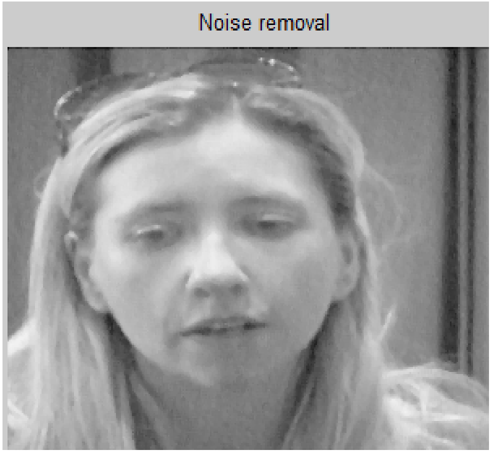

Figure 13 : Noise removals from video

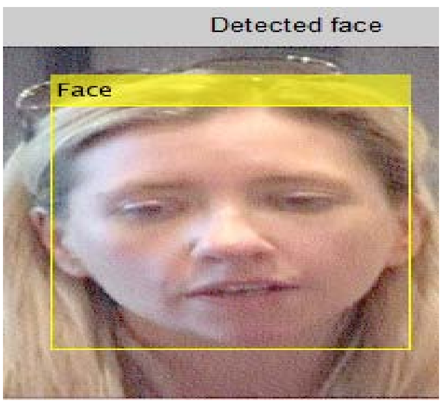

Figure 14 : Face detection

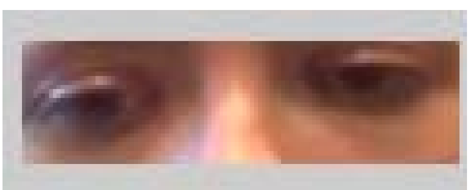

Figure 15 : Detected eyes

The figure 13 is the removal of salt and pepper noise from the input video using median filter and the figure 14,15 shows the face and eyes detected from the input video using viola Jones algorithm.

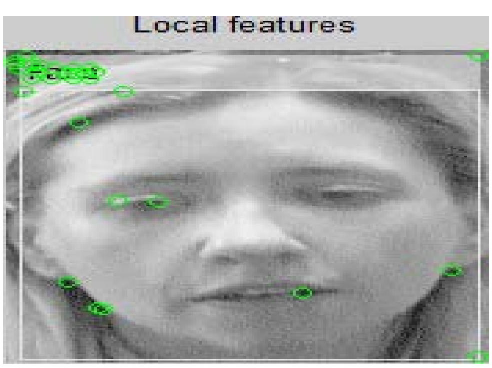

Figure 16 : Feature extraction

The figure 16. Shows the feature extraction process where the features of the eyes are extracted using Iterative randomized hough transform.

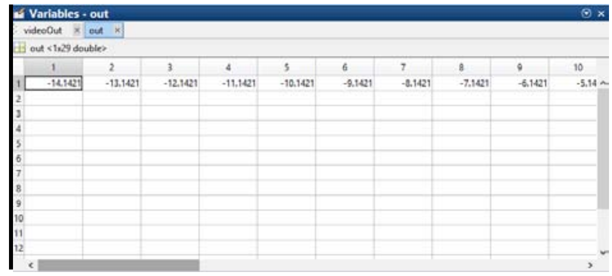

Figure 17 : IRHT command window

Figures 18 : Authentication

Figure 17. shows the output obtained from the features of the eyes using IRHT method and in the figure 18. Authentication of iris features can be obtained from IRHT.

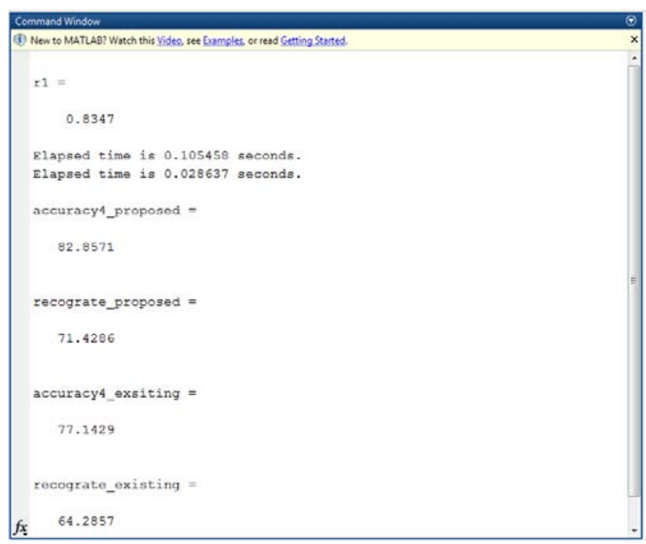

Figure 19 : Command window of mat lab by executing the program coding

The figure 19. gives the values needed for performance evaluation between adaptive matching and adaptive hamming distance.

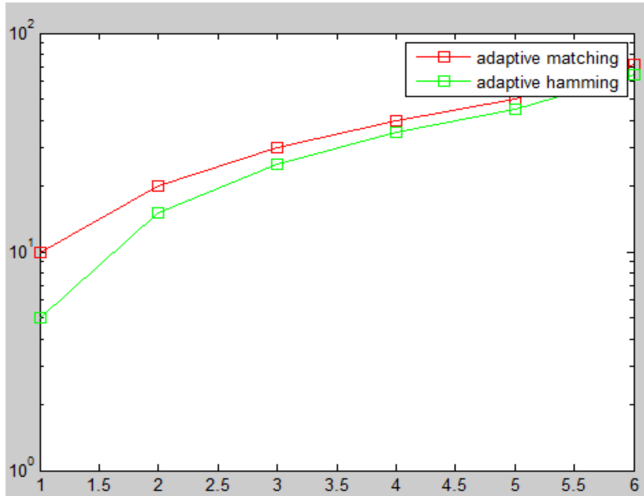

Figure 20 : Performance graph of adaptive matching versus adaptive hamming

The figure 20. Shows the comparison between accuracy and recognition rate of adaptive matching and adaptive hamming distance method.

c) Comparison table for face recognition method

Table 1 : Comparison chart of output values of face recognition

Algorithm	Accuracy	Recognition Rate
Template Matching (Existing)	93.33	30
DBPNN (PROPOSED)	96.66	33.33

d) Comparison table for iris recognition method

Algorithm	Accuracy	Recognition Rate
Adaptive Matching (Existing)	77.14	64.20
Adaptive Hamming Distance (Proposed)	82.55	71.42

Table Comparison chart of output values of Iris recognition

Recognition Rate is calculated by,

$$Recognition\ \ Rate = \frac{Total\ no.of.images}{correctly\ accepted\ images}$$

Accuracy is calculated by,

$$Accuracy = 100 \times True\ \ Positive\ \ rates$$

Angle of the face is calculated by,

$$Angle = obtained\ value\ \times 180$$

VI. CONCLUSION

Face recognition is a complicated task that requires efficient handling of complex variations in facial appearance caused by a range of factors such as, illumination variation, expressions, and aging effects. In the existing method, a highly compact and discriminative feature descriptor and adaptive matching framework is used for enhanced face recognition. But the method is not suitable for the face in side pose or different angle. In order to overcome this, a new methodology is proposed which performs face recognition under the combined effects of deblurring and denoising. In this method, Viola-Jones algorithm is initially used for detecting the face and dense feature extraction is used for feature extraction. DBPNN is used to detect the face from side pose and finally Iris detection is done by using Iterative randomized Hough transform were the pupil of the eyes is detected with number of iteration. The proposed method significantly reduces the dimension of feature representation and improved computational efficiency without sacrificing recognition performance. The future work is to recognize the face in side pose even from occluded face.

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AC Voltage Analysis using Matrix Converter

By Anubhab Sarker
American International University

Abstract- The purpose of this thesis is to design a three phase converter, whose switching pattern is arranged in a Matrix form and the converter is commonly known as three phase Matrix converter. This AC-AC system is proposed as an effective replacement for the conventional AC-DC-AC system which employs a two-step power conversion. The thesis analyzes the performance of matrix converter with two modulation techniques such as SVPWM and SVM. The basic principle and switching sequence of these modulation techniques have been presented. The output voltage, output current waveforms, voltage transfer ratio and THD spectrum of switching waveforms connected to different type of loads are analyzed by using Matlab/Simulink software.

GJRE-F Classification : FOR Code: 290901p

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I. Introduction

Matrix converters are capable of AC/AC direct power conversion. It does not have any dc-link circuit and does not need any large energy storage elements. The key element of a Matrix Converter is the fully controlled four quadrant bidirectional switch, which allows high frequency operation. The matrix converter consists of 9 bi-directional switches that allow any output phase to be connected to any input phase. space-vector modulation technique is used in Matrix converter modulation technique. The SVM technique was adapted for the matrix converter by employing a basic concept of indirect modulation using a fictitious DC bus, then dividing the converter into a rectification stage and an inversion stage. Furthermore, this modulation technique allows simplifying a converter model, making it easier to control the converter under imbalanced and distorted power supply conditions. BY using this technique Matrix converter generate variable frequency.

II. Literature Review

This part of the thesis consists of the details of modulation technique and the switching topology of matrix converter. The working principal of most of the technique will also be explained. The main technique those are used for analysis Matrix converter are:-

1. Space vector modulation
2. Modulation technique of Matrix converter
3. Matrix Converter Switching States
4. Topologies of Bi-directional Switches

Space vector modulation: Space vector modulation is an algorithm for the control of PULSE WIDTH MODULATION (PWM). It is used for the creation of ac wave forms. It is a general technique for any three-phase load, although it has been developed for motor control. Space vector pulse width modulation is applied to output voltage and input current control. This method is an advantage because of increased flexibility in the choice of switching vector for both input current and output voltage control. It can yield useful advantage under unbalanced conditions. The three phase variables are expressed in space vectors. For a sufficiently small time interval, the reference voltage vector can be approximated by a set of stationary vectors generated by a matrix converter. The modulation process thus required consists of two main parts: selection of the switching vectors and computation of the vector time intervals. SVPWM refers to a special switching sequence of the upper three power transistors of a three-phase power inverter. It has been shown to generate less harmonic distortion in the output voltages and or currents applied to the phases of an AC motor and to provide more efficient use of supply voltage. There are two possible vectors called zero vector and Active vector.

Modulation technique of Matrix converter: Matrix Converter operation can be explained in more general terms using a space vector approach. For operation of the Matrix Converter one and only one switch in each output phase must be conducting. This leads to twenty seven possible switching combinations for the Matrix Converter. By applying Equations 1 and .2 to determine the output voltage and input current vectors respectively, the magnitude and phase of these vectors for all possible combinations are needed. To find current and voltage modulation index power balance condition can be used. With balanced output load current condition such as,

$$\overline{V_{(it)}} = (qVim\sqrt{3}) \cos(\omega ot) \qquad (1)$$

$$\overline{V_{(0t)}} = \frac{2}{3}(v1o + av2o +)a^2 v3o \qquad (2)$$

Where, $a = e^{j\frac{2\pi}{3}}$

The vector $\overline{V_{(0t)}}$ has a constant length of qVin√3 and it is rotating at frequency ωo. The basis of the space vector modulation technique is that the possible output voltages for the converter can be expressed in the same form as Equation 2. At each sampling instant, the position of $\overline{V_{(0t)}}$ is compared with the possible vectors and the desired output voltage is synthesized by time averaging (within the switching interval) between

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adjacent vectors to give the correct mean voltage. For a conventional DC link inverter this process is very simple because there are only eight possible switching. The situation with a Matrix Converter is more complex as there are twenty seven possible switching states and the input voltages are time varying.

Matrix Converter Switching States: In matrix converter 27 switching combination is taken place. 18 combinations where the output voltage and the input current vectors have fixed directions with magnitudes that vary with the input voltage phase angle and the output current phase angle respectively.3 combinations giving null output voltage and input current vectors. All three output phases are connected to the same input phase in these combinations. Space vectors of output take one of 6 fixed positions (varying amplitude), Space vectors of input current take one of 6 fixed positions (varying amplitude). All space vectors are at the origin (zero length). 6 combinations in which each output phase is connected to a different input phase. Both magnitude and phase of the resultant vectors are variable in these cases. By this way total 27 switching combination is taken place in Matrix converter.

Topologies of Bi-directional Switch: Bi-directional switches capable of blocking voltage and conducting current in both directions are required by the Matrix Converter. Common Emitter Bi-directional Switch is used in this thesis which consists of two diodes and two IGBTs those are connected in anti-parallel. The diodes are included to provide the reverse blocking capability. The reverse blocking capability is a weak of the early IGBT technology . The advantage is that it is possible to independency control the direction of the current.

III. Designing of Simulink Models and Outputs

This part is carrying the whole design of Space vector modulation and Matrix converter models and analysis their outputs.

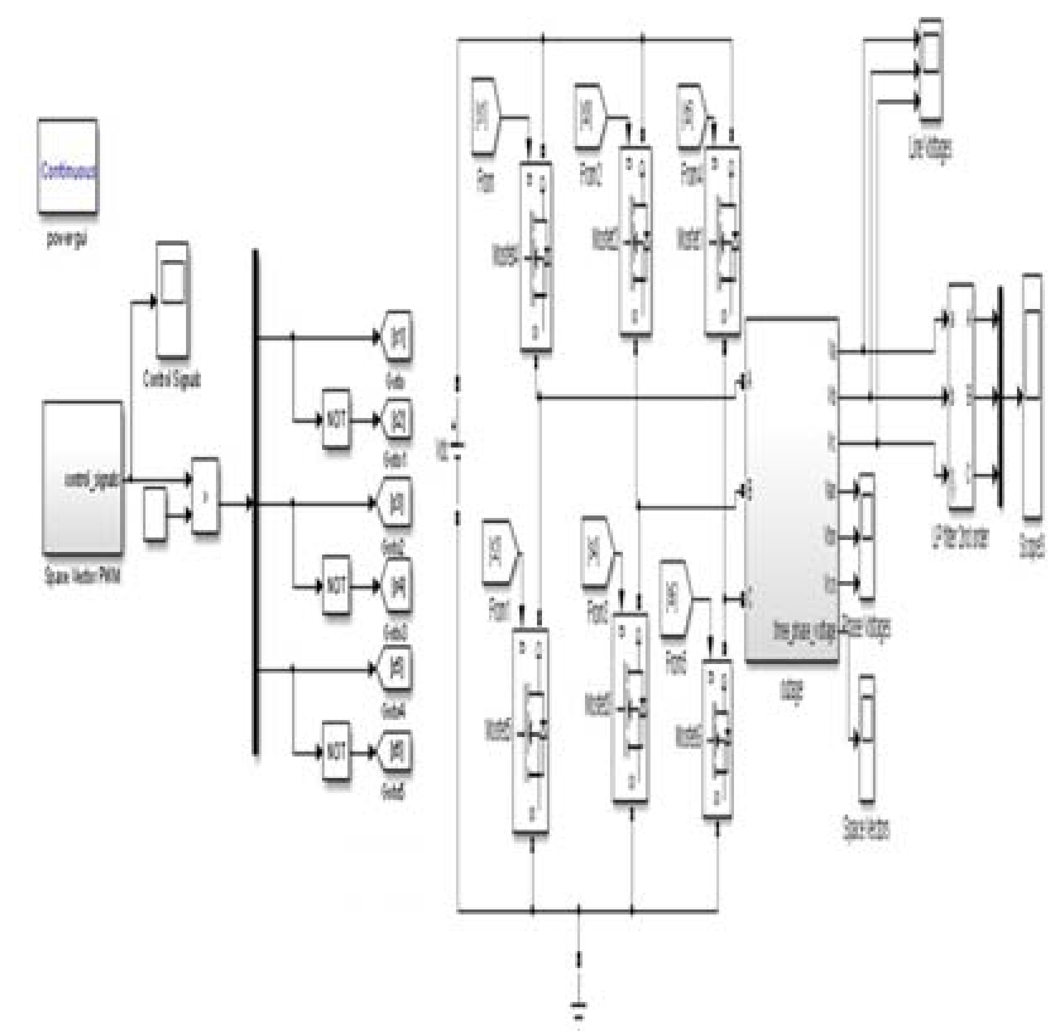

Figure 1 : Space vector modulation process

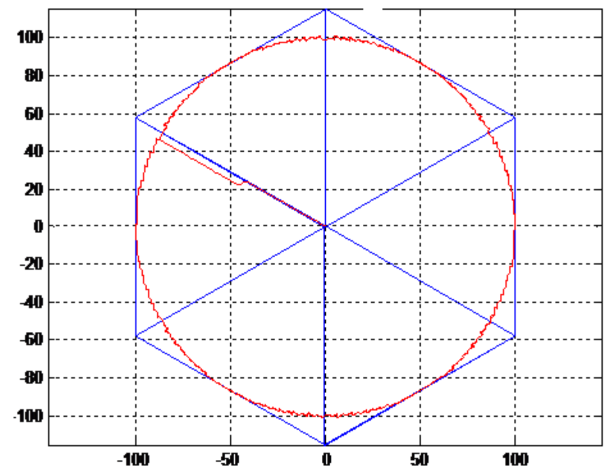

Figure 2 : space vector trajectory

Space vector pulse width modulation is applied to output voltage and input current control In figure 2 six sectors are shown for six switches. The objective of space vector PWM technique is to approximate the reference voltage vector Vref using the six switching patterns. Two switches from the same phase should not be on at the same time, otherwise short circuit will occur in the same input side. One switch is always on at different phase with Vref. So a continuous flow of bus voltage is shown in circle(red marked).

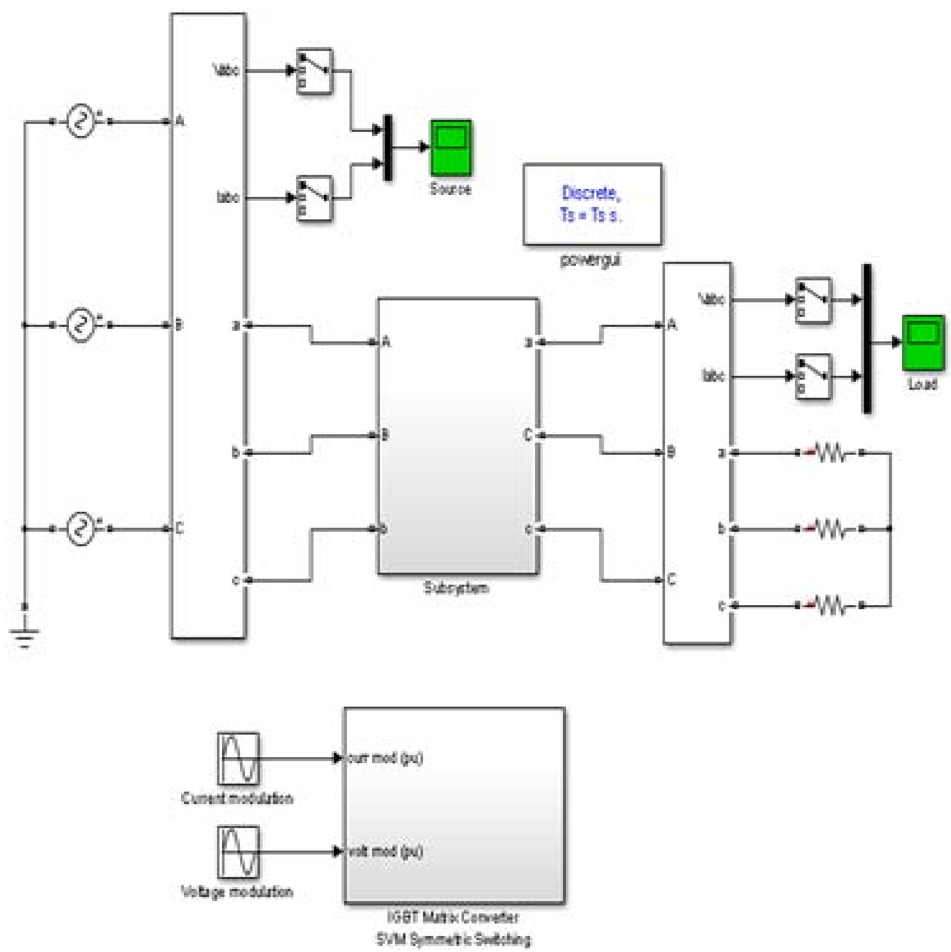

Figure 3 : General model of Matrix converter

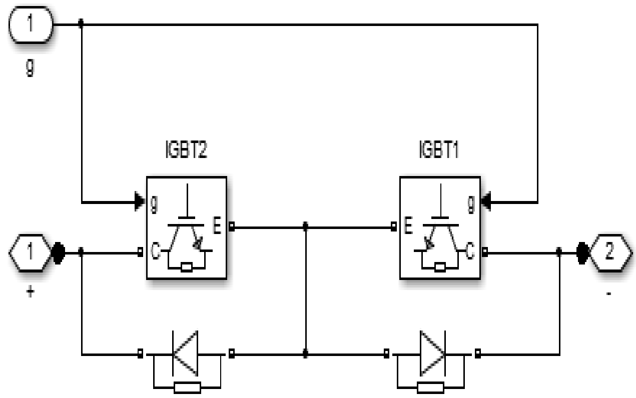

Figure 4 : Bidirectional switch

In figure 3 3ϕ supply is provided where the frequency is used 60Hz. This fundamental frequency is modulated by SVM technique. In this technique high switching frequency is needed for triggering. In figure 4 common ammeter bidirectional switch is used where the modulated signal is used as input signal. After finishing all the switching combination inside the SVM symmetric switching part the desired frequency will be gotten.

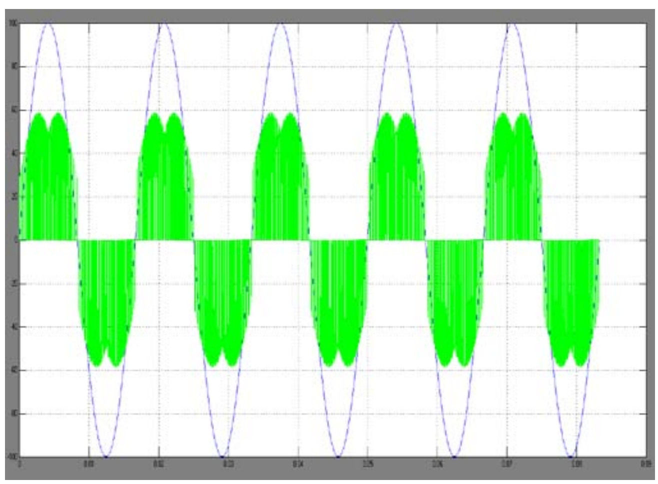

Figure 5 : Input wave shape of Matrix converter

Where, input voltage=100V(P-P)
Time period T=0.0168sec (approx)
And frequency f=1/T=59.82Hz

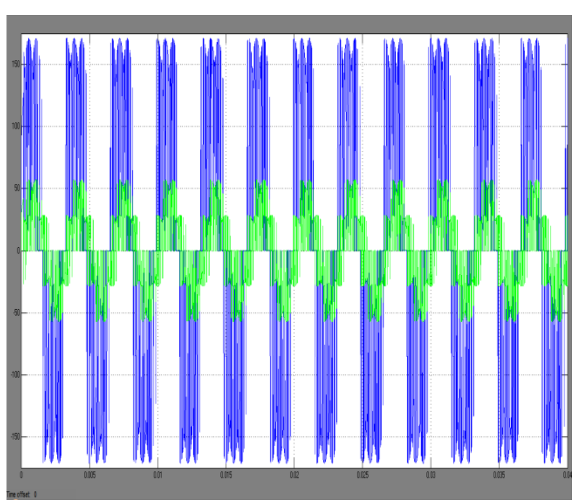

Figure 6 : Wave shape across the load

From the Output wave shape of the matrix converter, desired frequency is gotten. The input frequency is approximately 60hz but the output frequency is almost 333Hz. that's why it is called the unlimited frequency changer. And voltage also increased 160v (p-p).

Here, $T_{1/2} = 0.002$

So Frequency, F = 333.33 Hz

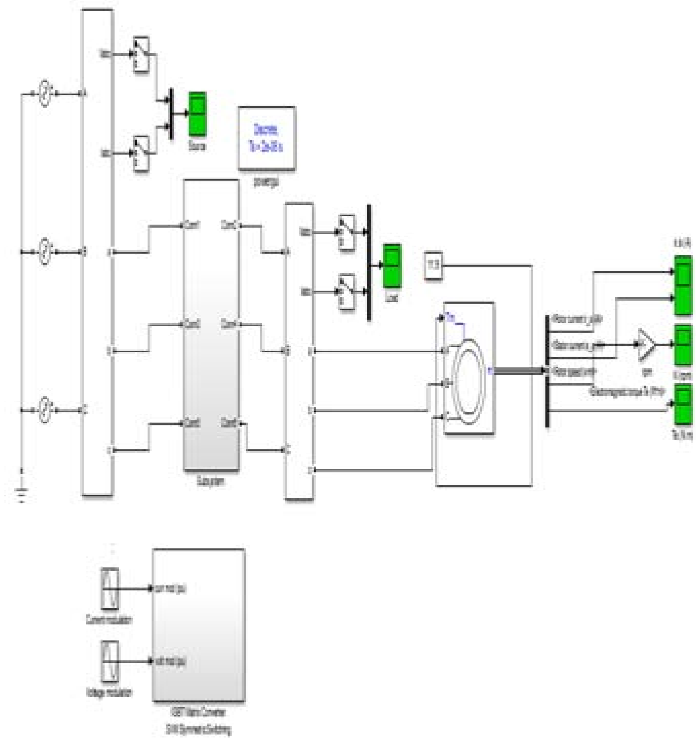

Figure 7 : Using induction machines as a load

A 220v, 60Hz asynchronous machine is used here as a load of Matrix converter in figure 7. The basic concept of an induction motor is an AC electric motor in which the electric current in the rotor needed to produce torque is obtained by electromagnetic induction from the magnetic field of the stator winding. Connecting this machine across the Matrix converter in figure 7 the stator current, rotor current, torque and the speed will be checked.

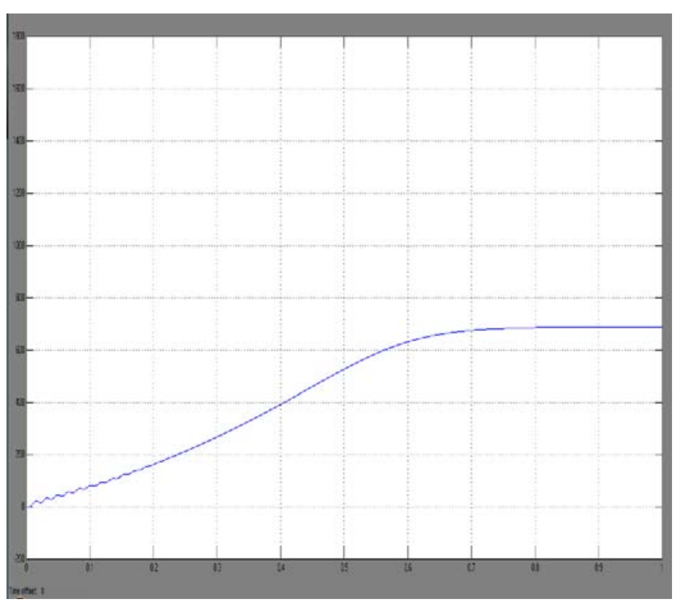

Figure 8 : induction motor speed

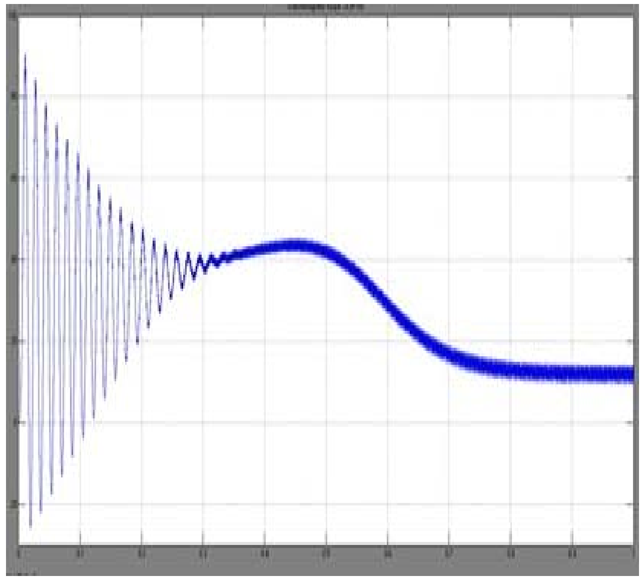

Figure 9 : Graphical representation for torque

In figure 9 it is shown that the motor speed is going 0 to 700rpm. in figure 10 electromagnetic troque is represented in a graph. A noisy level is observed because the stator fed by PWM.

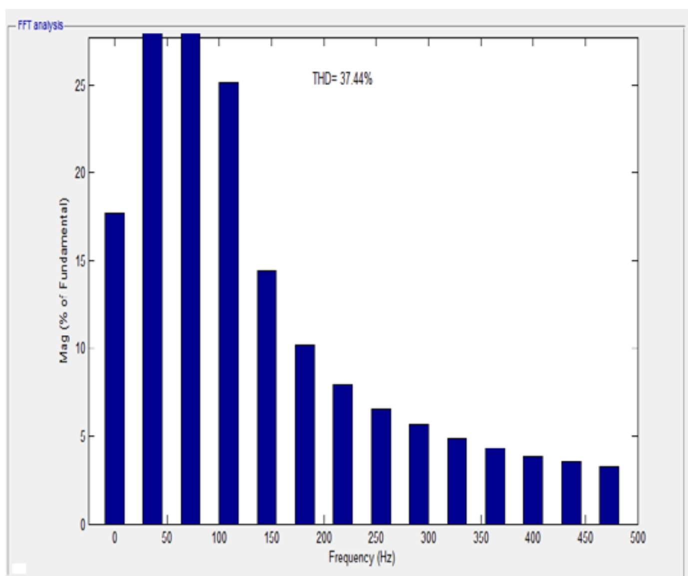

Figure 10 : THD analysis using FFT

When pulse signal is converted into ac signal some harmonic distortion is created that should not be cancelled. The lower the percentage, the higher the speed performance. It is related to frequency.

Calculation:

For a 4 pole induction machine:

Synchronous speed Ns = (120*f)/P

= (120*60)/4

Thus N_s= 1800 rpm

Since slip for a normal induction motor ranges between 0 and 1, S_{Tmax}=0.5

N_{Tmax} = (1-s)N_s

= (1-0.5)1800

N_{Tmax} = 900 rpm

But due to some extra harmonics, we are getting 700 rpm.

IV. Conclusion

The conversion from ac to ac is quite difficult. but matrix converter makes it simple and easy. It has appeared as an alternative solution for adjustable speed AC drive applications. This thesis presented easier methods for implementing complex switching strategies, studying and mitigating the effects of unbalance, and topological changes to increase the performance indices. The thesis also suggests modulation techniques to eliminate the common mode voltage and a new direct torque control procedure for controlling an induction motor fed by the modified matrix converter topology. The work also introduces a simple carrier based modulation technique, termed as the SVM technique, as an alternative way of implementing the space PWM technique for the matrix converter. Based on the analysis carried out on the original SV PWM technique, the thesis proposes a modified control algorithm. This modified algorithm reduces the input current harmonic distortion without affecting the output side performance.

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Intelligent Solar based Gesture Controlled Wireless Wheelchair for the Physically Handicapped

By Mayank Shadwani, Sachin Singh, Vinay Kumar Verma & Shivani Sachan
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Abstract- This document presents a wide new way of handling a wheelchair. The wheelchairs which were being used in the present time are controlled by the persons sitting over them. The person sitting over the chair has to exert a force by his hands in order to move the chair and get to the desired location. So for the sake of the handicapped persons, a new implementation is going to be made on the wheelchair which enables the person to move the chair without applying any kind of force. The person just has to make a simple gesture using his body part (for example: hand) to move the chair. This application is done using the microcontroller and the accelerometer sensor. The physically handicapped people will have the option of controlling the system through hand gesture wirelessly from ranges up to several meters and will have the independence of using the wheelchair without the help of any other people. The project includes the locomotion of the wheelchair without the application of any external force. This application is done using triple axis accelerometer sensor and the complete phenomenon is made wireless by using RF modules.

Keywords: *wireless control, gesture control, solar energy, distress call system (DCS).*

GJRE-F Classification : *FOR Code: 090605*

INTELLIGENTSOLARBASEDGESTURECONTROLLEDWIRELESSWHEELCHAIRFORTHEPHYSICALLYHANDICAPPED

Strictly as per the compliance and regulations of :

Intelligent Solar based Gesture Controlled Wireless Wheelchair for the Physically Handicapped

Mayank Shadwani [α], Sachin Singh [σ], Vinay Kumar Verma [ρ] & Shivani Sachan [ω]

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Keywords: wireless control, gesture control, solar energy, distress call system (DCS).

I. Introduction

The project displays the feature of the wireless controlling of the objects. In the present project, a very new way of handling a wheelchair by the application of the accelerometer sensor. We all know that while driving the wheelchair the person has to exert a force on the wheels so that the wheelchair can move in the desired direction. To achieve this performance the application involves the use of the *triple axis accelerometer sensor (ADXL335).* Also for the controlling purpose, microcontroller is employed in the circuit. People with physical disabilities and partial paralysis always find it difficult to navigate through their habitat or their home without the assistance of someone. Often after paralysis or physical disability the wheelchair is the most common means of locomotion for such people. But to navigate through one's own house without help of someone every time can be demoralizing for the person as well.

With the present development on the field of robotics, embedded system and artificial intelligence a successful project has been developed in order to easily solve this matter and that too at a very low cost. The wheelchair in context can be remotely controlled from several meters wirelessly without actually sitting on it. The chair can be controlled by hand gesture method with directions as needed. The recognition of hand gestures requires both hand's detection and gesture's recognition. Both tasks are very challenging, mainly due to the variability of the possible hand gestures (signs), and because hands are complex, deformable objects (a hand has more than 25 degrees of freedom, considering fingers, wrist and elbow joints) that are very difficult to detect in dynamic environments with cluttered backgrounds and variable illumination.

Several techniques have been discovered so far to trace the position of the hands and turn them into the corresponding voltage level. The standard input methods, such as text input via the keyboard and pointer/location information from a mouse, do not provide a natural interaction between humans and machines. Therefore, it is essential to create models for natural communication between humans and machines. The ability to understand hand gestures will improve the naturalness and efficiency of human interaction with machine, and allow the user to communicate in complex tasks.

The proposed model makes the wheelchair a lot easier to assemble and simple in the use, in addition the cost of manufacturing also gets reduced. Also, with this development the wheelchair can be controlled remotely from several meters away. So a person sitting on the

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sofa can control the wheelchair near or away from him just by hand gestures. It can also help people during the night without the need for a third person; for the person to get on the wheelchair and move inside the house. People will also be able to control the chair in narrow spaces without collision as the system uses ultrasonic sensors and proximity sensors to avoid the objects. This technique of avoiding the obstacles in the path permits the person sitting on the wheelchair to move freely without having any tension of falling down or striking to any wall or obstacle. The only reason behind developing this model is to make handicapped peoples tension free. They don't require anyone's help to go anywhere. All what they have to do is to give command to the chair using their working body parts. Also a great advantage is that this implementation will add just a little cost in the original manufacturing cost of the wheelchair, so that anyone can afford this wheelchair.

II. Block Diagram

The main aim of the project is to allow the physically challenged peoples to move the wheelchair without the help of anyone. Because of this development such peoples can easily move anywhere in their homes without having any fear of getting struck by any obstacle. This model consists of controlling the wheelchair with the use of the gesture sensors and the microcontroller unit. The output from the hand gestures modules gives command to the microcontroller. The whole technology is made wireless just by employing a transmitter and a receiver module which gives a reduction in the complexity of the circuit arising because of the wires. For wireless communication RF modules are used. The following block diagram shows how the entire system working:

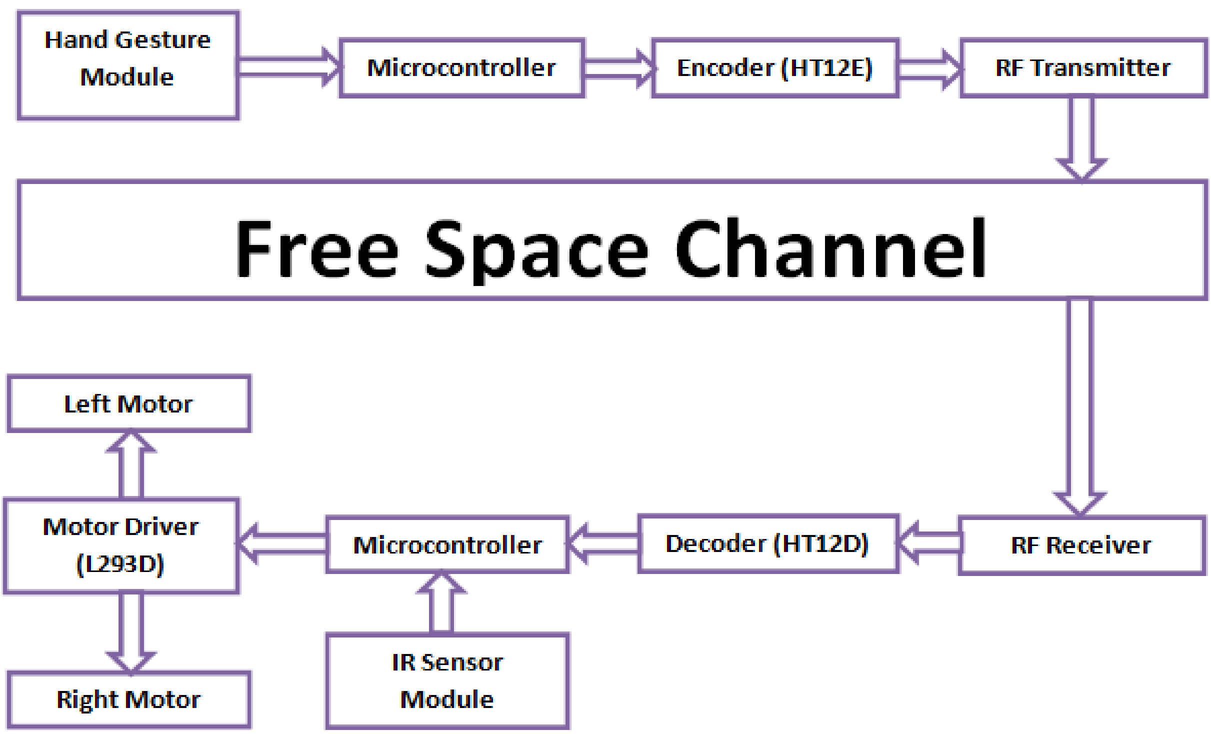

Figure 1 : Block diagram of the system

As the block diagram shows, the input signal is received from the accelerometer sensor, and then the output of the sensor is given to the RF transmitter module. As we are concerned with the wireless technology, so here we have employed RF modules. The transmitted wave travels through the channel and reach to the receiver module. The receiver gives the output which is originally fed to the system. The output of the system is given to the microcontroller which is mainly responsible for the entire controlling phenomenon. Here we have used PIC microcontroller. Motion control, edge detection and obstacle avoiding processes are controlled by the microcontroller. Now we will move towards the discussion of the components used in the system.

III. System Hardware

The system uses various components which are important for the system to work in the proper manner. The description about the various components is as follows:

a) Solar Panel

Solar energy is used to give supply to the whole system. As we are concerned with the affordable use of the system so we have employed solar energy instead of the main electricity. A battery of 12V 1.5AH is used for the backup so that the chair can work in night time also. The charging circuitry includes the switching diode (1N4148) in parallel to the capacitor. In addition to this a 15V regulator IC (7815) is also used to regulate the voltage coming from the panel. The output of the regulator IC is given to the battery.

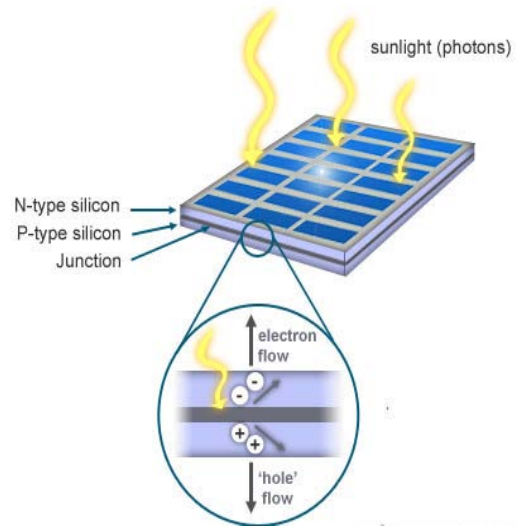

Figure 2 : Working of a solar panel

Figure 3 : Battery used for backup purpose

b) Hand Gesture Module

The module which is being used here for detecting the gestures made by hand is a *triple axis accelerometer sensor (ADXL335)* which measures the three dimensional position of the hand. The output from the accelerometer is an analog voltage which is further converted in the corresponding digital voltage with the help of ADC (Analog to Digital Converter).

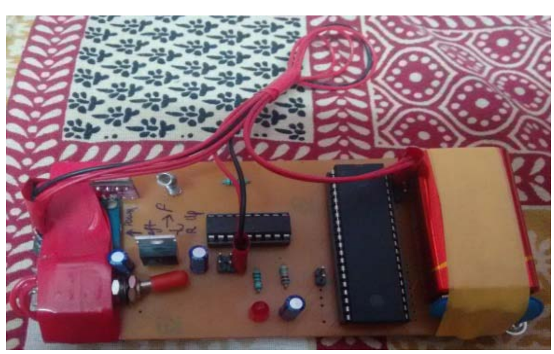

Figure 4 : View of the Hand Gesture Module used for controlling the locomotion

The accelerometers are used in many applications and are of many types depending upon the axis. The accelerometer sensor senses the accelerating force (acceleration due to gravity or g) and thus gives a particular voltage for the x, y and z coordinate orientation. The data can be observed in integer format through the serial port of MCU (Microcontroller Unit) on the computer's serial monitor and accordingly the orientations of the hand can be sorted out. Accelerometer sensors convert either linear or angular acceleration to an output signal. Accelerometer sensors use Newton's second law of motion.

$$\text{Force} = \text{mass } \times \text{acceleration}$$

If the force and mass on the body are known, then we can calculate the accelerating force acting on the body. Thus, by this equation the acceleration due to gravity is determined. As we know that acceleration is defined by the rate of change of velocity per unit time.

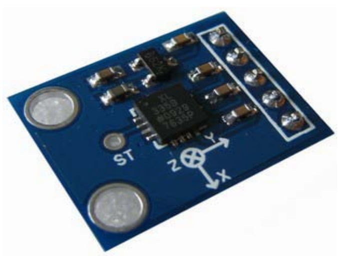

Figure 5 : Accelerometer Sensor (ADXL335)

c) Microcontroller

Microcontroller is such a thing which is used to control any system according to the requirement of the user. Microcontroller is rather different from the microprocessor. In the microcontroller, there is no option to attach external memories to the devices. Since we are concerned with the small application and didn't require large memory so here we have employed a microcontroller. Microcontrollers are of different series and need to be programmed as per the requirement of the application. The different classes of MCU available in the market are: **8051, AVR,** & **PIC.** Here in this project we have used AVR series MCU. Microcontroller is the main device which is used to control the entire phenomenon.

d) Encoder & Decoder

Encoder (HT12E) and decoder (HT12D) are used in the system for the purpose of encoding the signal at the transmitter end and decoding the signal at the receiver end. The encoder generates serial codes that are automatically sent three times and must be received at least twice before data is accepted as valid by the decoder circuit.

e) Radio Frequency Module

The RF transmitter module has been used as per the purpose of making the gesture module completely wireless. We are using a 434MHz WS-TX-01 module for the transmitting purpose. For a specific orientation of the hand the MCU on the hand decides the condition and a particular character are sent to receiver module. For the receiver we are using a 434MHz WS-RX-02 module which is a low cost receiver module. The receiver upon receiving the string sends the data to the microcontroller on the wheelchair which in turn decides the case of the locomotion of the wheelchair. The RF module works on the principle of Amplitude Shift Keying (ASK) modulation technique. The entire module is very effective for long distances.

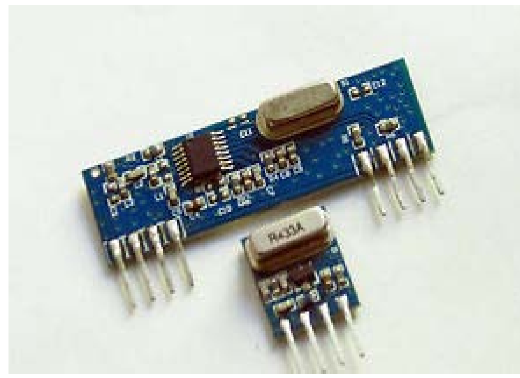

Figure 6 : 433.92MHz RF Modules (Transmitter & Receiver)

f) Locomotion

For the locomotion purpose we are using a L293D motor driver IC. As the output current of microcontroller is low and it is not sufficient to drive the motor so driver IC is used. ULN2003 can also be used for this purpose. The motors are controlled by the bidirectional motor driver IC. The motor driver is connected through the microcontroller on the wheelchair which sends the signal to the driver for various conditions. For smooth turning during the motion of the wheelchair we have used the method of PWM. The Pulse Width Modulation (PWM) allows the microcontroller to send the power to the motor driver in small packets over high frequency. The constant ON and OFF states actually helps to conduct smooth turning operation.Motor drivers act as current amplifiers since they take a low-current control signal and provide a higher-current signal. This higher current signal is used to drive the motors.

g) Edge Detection Module

For this purpose we have employed IR sensors module to avoid the edges. An Infrared (IR) sensor is used to detect obstacles in front of the chair or to differentiate between colours depending on the configuration of the sensor. An IR sensor consists of an emitter, detector and associated circuitry. The emitter is simply an IR LED (Light Emitting Diode) and the detector is simply an IR photodiode which is sensitive to IR light of the same wavelength as that emitted by the IR LED. When IR light falls on the photodiode, its resistance and correspondingly, its output voltage, change in proportion to the magnitude of the IR light received. This is the underlying principle of working of the IR sensor.

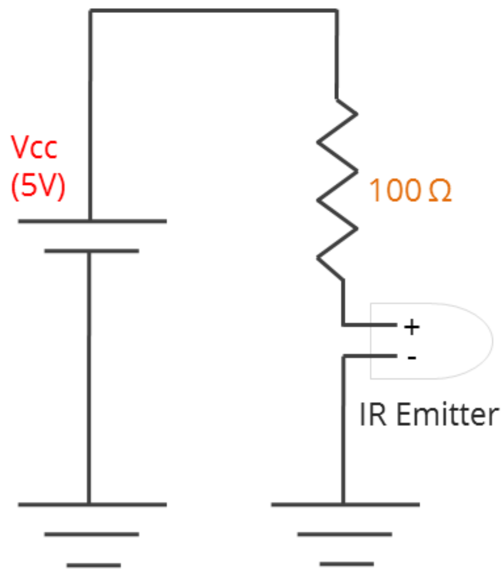

Figure 7 : Emitter Circuit

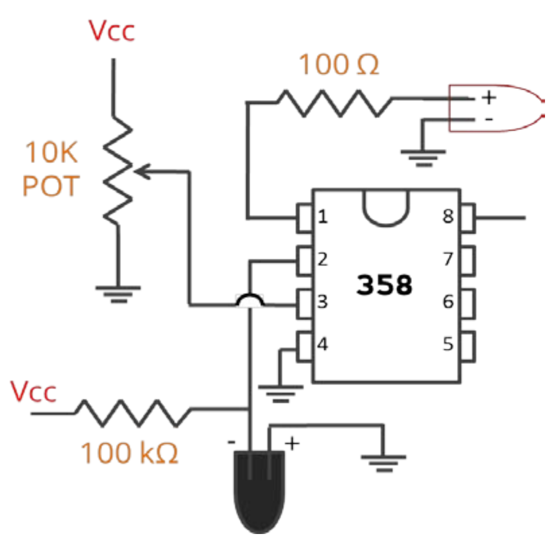

Figure 8 : Detector Circuit

h) Distress Call System

For the case of emergency of the handicapped person an emergency call system is used by implementing the circuit with the SIM module. The distress call system is used along with the gesture command system. On the sensation of malfunction or if there is an emergency situation the person can send an SOS distress message by GSM controlled SIM 900 module on the wheelchair. The hand module in gesture control has a switch which if pressed, sends a digital high signal to the MCU on the hand which in turns sends the signal wirelessly to the receiver module on the wheelchair and then to the SIM900 module. The module then send the distress SMS to the already inserted numbers of people at the same time.

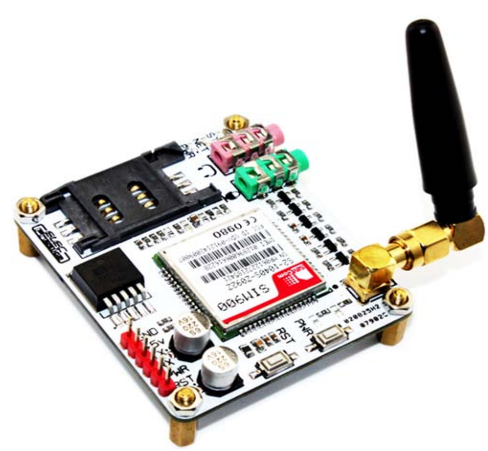

Figure 9 : GSM900 Module

IV. Results and Discussions

All the components after integration give us the working skeleton model for the wheelchair. The wheelchair model works perfectly according to the hand gestures. The reaction time of the various modules as noticed after the several trial runs are tabulated below. The readings are calculated based on the following formula:

*Success Rate = (Number of successful trails * 100) / Total number of trials*

Along with that we also noticed that the communication of the wireless module extends over several meters in the indoors. Rather than the constricted need of the person to be on the wheelchair the entire time to control it, the developed idea works absolutely free of any wires or any restrictions. The ability to avoid the obstacles is kept to the limit of 5cm from the wheelchair. If detected the wheelchair turn the other direction from the obstacle still it is back on the front line. The various gestures were tested and the outputs were studied to check if the right codes are transmitted. The mobile wheelchair is capable of sending over the air how close it was to the nearest object at its right, left and centre.

Table1 : Results obtained from observations

S.No.	Parameters	Success Rate
1.	Gesture Change	94%
2.	Distress Alert	100%
3.	Edge Detection & Avoiding	100%

V. Final View of the Prototype

Figure 10 : View of the prototype made for testing different parameters

V. Conclusion

With the development of the project it can be successfully implemented on a larger scale for the handicapped people. The low cost of the assembly makes it really a bonus for the general public. The wireless system will be a boost to the confidence and willpower of physically challenged people as it will help them to be self-reliable. As a part of further development the project can be developed with addition voice recognition features. There can also be the application of intelligent home navigation for handicapped people to go through the entire house and get help from technological interface for the navigation. The object avoiding and careful navigation principle can be improved with algorithm based image processing technology.

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Identification of Premature Ventricular Contraction (PVC) of Electrocardiogram using Statistical Tools and Non-Linear Analysis

By Farhana Akter Mou, Effat Jerin, Md. Abdullah Al Mahmud & A.H.M Zadidul Karim

University of Asia Pacific (UAP)

Abstract- Non-linear analysis is a useful technique in a medical field specially in cardiac cases. Statistics tools & Non-linear parameters have shown potentiality to the identification of diseases, especially in the analysis of biomedical signals like electrocardiogram (ECG). In this work, premature ventricular contraction (i.e abnormality) in ECG signals has been analysed using various non-linear techniques. First, the ECG signal is processed through a series of steps to extract the QRS complex. From this extracted feature, bit-to-bit interval (BBI) and instantaneous heart rate (IHR) have been calculated.

Index Terms: electrocardiogram (ECG), premature ventricular contraction (PVC), instantaneous heart rate (IHR), standard deviation(SD), central tendency measure (CTM).

GJRE-F Classification : FOR Code: 090699

IDENTIFICATIONOFPREMATUREVENTRICULARCONTRACTIONPVCOFELECTROCARDIOGRAMUSINGSTATISTICALTOOLSANDNONLINEARANALYSIS

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Farhana Akter Mou [α], Effat Jerin [σ], Md. Abdullah Al Mahmud [ρ] & A.H.M Zadidul Karim [ω]

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Index Terms: electrocardiogram (ECG), premature ventricular contraction (PVC), instantaneous heart rate (IHR), standard deviation(SD), central tendency measure (CTM).

I. Introduction

a) Heart and ECG

The heart is the muscular organ that pumps the blood through the circulatory system by rhythmic contraction and dilation. In vertebrate there may be up to four chambers with two atria and two ventricles. Measuring the electrical activity of heart to show whether or not it is working normally and records the heart rhythm and activity on a moving strip of paper or a line on a screen, in a word that is called ECG. Electrocardiogram (ECG) is a wave that represents an electrical event in the heart, such as atrial depolarization, atrial repolarization, ventricular depolarization, ventricular repolarization, or transmission, and so on [1-4].

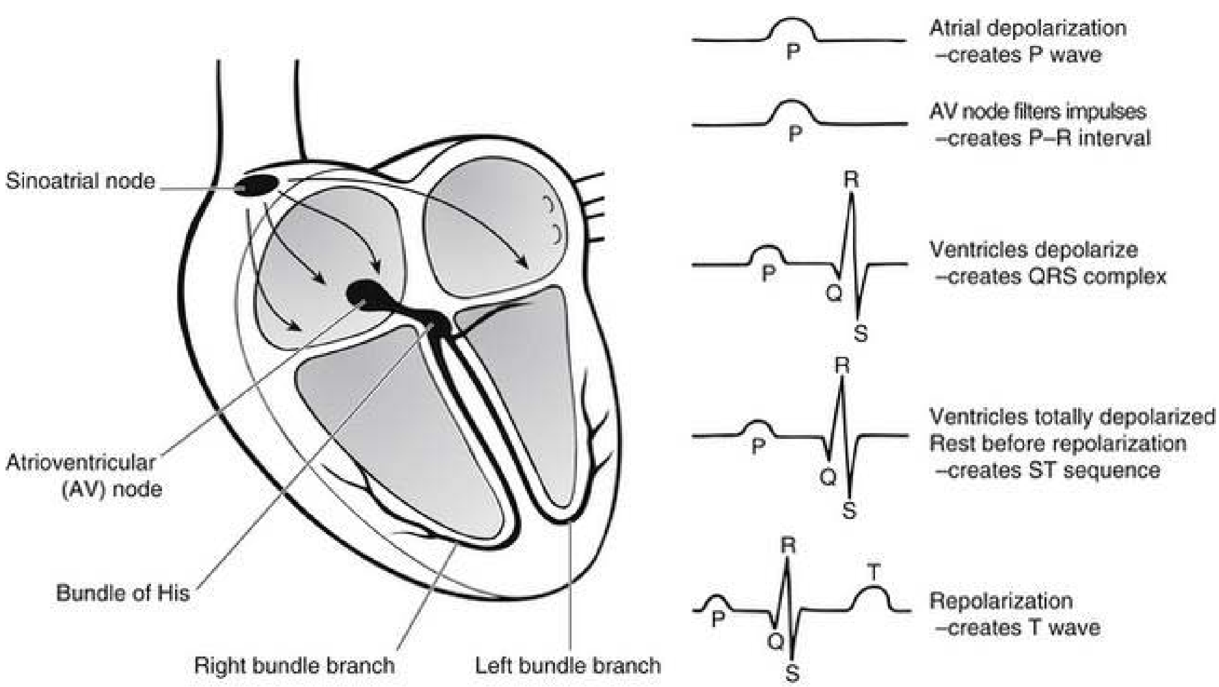

Figure: 1.1 : Anatomy of heart and ECG generation

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The electric current generated by depolarization and repolarization of the atria and ventricles is detected by electrodes, it is amplified, displayed on an oscilloscope, recorded on ECG paper, or stored in memory. The electric current generated by atrial depolarization is recorded as the P wave, and that generated by ventricular depolarization is recorded as the Q, R, and S waves: the QRS complex. Atrial repolarization is recorded as the atrial T wave (Ta), and ventricular repolarization, as the ventricular T wave, or simply, the T wave. The sections of the ECG between the waves and complexes are called segments and intervals: the PR segment, the ST segment, the TP segment, the PR interval, the QT interval, and the R-R interval. When electrical activity of the heart is not being detected, the ECG is a straight, flat line – the isoelectric line or baseline.

II. Proposed Method

This work presents heart rate variability (HRV) analysis using some non-linear methods. The ECG signal to be analyzed is first processed [5] to extract the QRS complex. From that bit-to-bit interval (BBI) is calculated. From the BBI the instantaneous heart rate (IHR) is found. On this dataset of BBI and IHR, various non-linear parameters like Poincare plot analysis (PPA), central tendency measure (CTM), phase space portrait, detrended fluctuation analysis are determined. The result is very effective to distinguish the ECG signals between the healthy person and that of the ailing person.

a) *Phase space portrait*

Phase space or phase diagram is such a space in which every point describes two or more states of a system variable. The number of states [6] that can be displayed in phase space is called dimension or reconstruction dimension. It is usually symbolized by the letter d or E. From the given digitized data x(1), x(2), …, x(n) of the IHR or BBI, a matrix A is obtained with its two columns given by x(1), x(2), …, x(n-τ) and x(1+ τ), x(2+ τ),.., x(n). Here τ is the time delay. The Phase space plot is constructed by plotting the data set with the time delay version of itself. The attribute of the reconstructed phase space plot depend on the choice of the value for τ. τ is measured through applying a autocorrelation function. Autocorrelation is a mathematical tool used frequently in signal processing for analyzing functions or series of values, such as time domain signals. Informally, it is a measure of how well a signal matches a with time-shifted version of itself, as a function of the amount of time shift. More precisely, it is the cross-correlation of a signal with itself. Autocorrelation is useful for finding repeating patterns in a signal, such as determining the presence of a periodic signal which has been buried under noise, or identifying the missing fundamental frequency in a signal implied by its harmonic frequencies. τ is typically chosen as the time it takes the autocorrelation function of the data to decay to 1/e or the first minimum in the graph of the average mutual information. Here we used the two dimensional phase space portrait, i.e., d = 2.

Here in this project, phase space analysis has been used on IHR time series and the results are analyzed to see if any significant difference is found between normal and abnormal data series.

Following are the portraits obtained using phase space portrait on IHR. They are presented along with the IHR plot against each sample.

b) *Poincare plot Analysis*

The most commonly used non-linear method of analyzing heart rate variability is the Poincare plot. The Poincare plot analysis (PPA) [7] is a quantitative visual technique, whereby the shape of the plot is categorized into functional classes and provides detailed beat-to-beat information on the behaviour of the heart. Poincare plots are applied for a two-dimensional graphical and quantitative representation where RR_j is plotted against $RRj+1$.Most commonly, three indices are calculated from Poincare plots: the standard deviation of the short-term RR-interval variability (SD1), the standard deviation of the long-term RR-interval variability (SD2) and the axes ratio (SD1/SD2) [8].

The standard deviation of the point's is perpendicular to the line-of identity denoted by SD1 describes short-term variability which is mainly caused by RSA. It can be shown that SD1 is related to the time-domain measure SDSD by,

SD12 =1/2 SDSD2................................. (2.1)

The standard deviation along the line-of-identity denoted by SD2, on the other hand, describes long-term variability and has been shown to be related to time-domain measures SDNN and SDSD by,

SD22 =2SDNN2-1/2 SDSD2....................... (2.2)

The standard Poincare plot can be considered to be of the first order. For the healthy heart, PPA shows a cigar-shaped cloud of points oriented along the line of identity.

In Poincare plot analysis here is the record of seven normal person's and seven abnormal person's ECG and analysis SD, MEAN VALUE, VAREANCE, RMSSD, SDSD.

Data table for SD, MEAN, VARIANCE, RMSSD & SDSD:

Table 2.1 : For normal patients

Patients name	SD	MEAN	VARIANCE	RMSSD	SDSD
107	5.6302	71.3858	31.6986	71.6073	6.2638
111	3.9393	70.7966	15.5179	70.9061	3.9646
112	2.5849	84.4500	6.6817	84.4895	2.3993
117	2.3313	51.0788	5.4351	51.1320	1.998
121	6.2172	62.3587	38.6533	62.6677	3.5439
122	4.5251	82.5183	20.4762	82.6422	2.2118
124	6.1893	54.3189	38.3076	54.6701	7.0596
AVERAGE	4.4881	68.1295	22.3957	68.3021	3.9202

Table 2.2: For Abnormal patients

Patients name	SD	MEAN	VARIANCE	RMSSD	SDSD
106	27.25	75.228	742.65	80.00	43.31
119	22.521	72.091	507.20	75.52	39.9055
208	19.397	101.93	376.244	103.75	30.7995
213	15.796	108.307	33.5941	108.462	8.8309
221	22.890	86.448	523.992	89.426	35.141
223	15.254	88.874	232.689	90.1735	25.6777
233	29.43	109.10	866.2365	113.001	50.6858
AVERAGE	21.79	91.711	468.9451	94.3464	33.4788

c) Detrended Fluctuation Analysis

Detrended Fluctuation Analysis is an interesting method for scaling the long-term autocorrelation of non-stationary signals [9-12]. It quantifies the complexity of signals using the fractal property. This method is a modified root mean square method for the random walk. Mean square distance of the signal from the local trend line is analyzed as a function of scale parameter. There is usually power-law dependence and interesting parameter is the exponent.

Detrended fluctuation analysis (DFA) measures the correlation within the signal. The correlation is extracted for different time scales. First, the RR interval time series is integrated,

$$(k) = \Sigma\, (RR_{j\ kj=1} - \overline{RR})\quad k=1,\ldots.N \qquad \ldots..(2.3)$$

Where $\overline{RR}$ is the average RR interval. Next, the integrated series is divided into segments of equal length n. Within each segment, a least squares line is fitted into the data. Let $y(k)$ denote these regression lines. Next the integrated series (k) is detrended by subtracting the local trend within each segment and the root-mean-square fluctuation of this integrated and detrended time series is calculated by,

$$F(n) = \sqrt{\frac{1}{N}\sum_{k=1}^{N}[y(k) - y_n(k)]^2} \qquad (2.4)$$

This computation is repeated over different segment lengths to yield the index (n) as a function of segment length n. Typically $F(n)$ increases with segment length. A linear relationship on a double log graph indicates presence of fractal scaling and the fluctuations can be characterized by scaling exponent **α** slope of the regression line relating log (n) to log n.

In the DFA method, the fractal-like signal (1/f noise) results in exponent value α =1.0, the white noise results in value 0.5, and the Brownian noise in value 1.5. White noise indicates a simulated uncorrelated random time series. The white noise is the value at one instant that does not correlate with any previous value, and the Brownian noise is the integration of the white noise. The 1/f noise can be interpreted as a "compromise" between the complete unpredictability of white noise and the much smoother "landscape" of Brownian noise.

Here DFA1 & DFA2 for the normal patients and abnormal patients are taken and plotted them.

Table 2.3 : Detrended fluctuation analysis α1 and α2 for normal patients

Patients name	107	111	112	117	121	122	124	AVERAGE
α1	0.835	0.789	0.603	0.845	1.143	1.140	0.406	0.823
α2	0.55	0.75	1.21	1.115	1.38	1.05	1.05	1.0270

Table 2.4 : Detrended fluctuation analysis α1 and α2 for abnormal patients

Patients name	106	119	208	213	221	223	233	AVERAGE
α1	0.321	0.216	0.200	0.404	0.342	0.265	0.245	0.2847
α2	0.776	0.359	0.619	0.207	0.674	0.457	0.233	0.4750

d) Central Tendency Measure

Central tendency measure (CTM) is used to quantify the degree of variability in the second order difference plot [13- 15]. It is calculated by selecting a circular region of radius r, around the origin, counting the number of points that fall within the radius, and dividing by the total number of points. If t = total number of points, and r = radius of central area.

Then,

$$n = \frac{[\sum_{i=1}^{t-2} \delta(di)]}{t-2} \quad (2.5)$$

Where,

$$\delta(di) = 1, \text{ if } [(a_{i+2} - a_{i+1})^2 + (a_{i+1} - a_i)^2]^{0.5} < r \quad (2.6)$$

$$= 0, \text{ otherwise}$$

In this paper, the data set from BBI and IHR are used to measure the CTM. In first step, the optimum radius is determined for the circular region.

It is determined the central tendency measures (CTM) with the variation of standard deviation (4 & 6) which varies from 10% to 100% for normal and abnormal patients. This is shown in table 2.5 to 2.8.

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Table 2.5 : CTM value for SD-4 varies from 10% to 100% obtained from IHR of normal rhythm

PATIENT NAME(CTM VALUE 0F SD)	107	111	112	117	121	122	124
10%	0.0084	0.0085	0.0197	0.0464	0.0361	0.0182	0.0260
20%	0.0436	0.0349	0.0868	0.1522	0.1659	0.0724	0.0959
30%	0.1072	0.0797	0.1759	0.2926	0.3561	0.1505	0.1975
40%	0.2013	0.1452	0.2892	0.4592	0.5334	0.2447	0.3038
50%	0.2921	0.2301	0.4158	0.6166	0.6920	0.3515	0.4016
60%	0.4003	0.3215	0.5250	0.7511	0.7953	0.4773	0.5043
70%	0.4939	0.4130	0.6355	0.8589	0.8766	0.5769	0.5947
80%	0.6011	0.5064	0.7239	0.9286	0.9294	0.6764	0.6863
90%	0.7065	0.5908	0.7893	0.9589	0.9542	0.7540	0.7550
100%	0.7804	0.6671	0.8430	0.9765	0.9725	0.8131	0.8137

Table 2.6 : CTM value for SD-4 varies from 10% to 100% obtained from IHR of abnormal rhythm

PATIENT NAME(CTM VALUE 0F SD)	106	119	208	213	221	223	233
10%	0.0015	0.0050	0	0.0037	0	0.0073	0.006
20%	0.0109	0.0181	0.0017	0.0246	0	0.0288	0.0072
30%	0.0247	0.0378	0.0615	0.0058	0.0003	0.0550	0.0156
40%	0.0400	0.0615	0.0095	0.0748	0.0004	0.0915	0.0231
50%	0.0647	0.0958	0.0136	0.1152	0.0012	0.1307	0.0322
60%	0.0954	0.1285	0.0221	0.1577	0.0025	0.1745	0.0494
70%	0.1215	0.1522	0.0283	0.2057	0.0029	0.2345	0.0605
80%	0.1546	0.1830	0.0337	0.2519	0.0054	0.2899	0.0778
90%	0.1873	0.2172	0.0426	0.3021	0.0066	0.3148	0.0888
100%	0.2184	0.2394	0.0525	0.3548	0.0083	0.3902	0.0996

Table 2.7 : CTM value for SD-6 varies from 10% to 100% obtained from IHR of normal rhythm

PATIENT NAME(CTM VALUE OF SD	107	111	112	117	121	122	124
10	0.01 97	0.01 74	0.03 87	0.08 75	0.08 57	0.03 72	0.05 14
205	0.10 72	0.07 97	0.17 59	0.29 26	0.35 61	0.15 05	0.10 74
30%	0.24 86	0.18 58	0.35 11	0.53 82	0.62 12	0.30 34	0.35 58
40%	0.40 03	0.32 15	0.52 50	0.75 11	0.79 53	0.47 73	0.50 43
50%	0.54 73	0.46 11	0.68 64	0.90 07	0.90 52	0.62 90	0.63 80
60%	0.70 65	0.59 08	0.78 93	0.95 89	0.95 42	0.75 40	0.75 50
70%	0.80 76	0.69 59	0.86 82	0.98 37	0.97 95	0.83 94	0.83 85
80%	0.86 80	0.77 51	0.92 19	0.99 02	0.98 81	0.90 49	0.88 68
90%	0.89 04	0.85 01	0.95 98	0.99 22	0.99 19	0.94 42	0.92 02
100%	0.89 51	0.90 43	0.97 95	0.99 41	0.99 19	0.96 72	0.93 75

Table 2.8 : CTM value for SD-6 varies from 10% to 100% obtained from IHR of abnormal rhythm

PATIENT NAME(CTM VALUE 0F SD)	106	119	208	213	221	223	233
10%	0.00 69	0.01 11	0.0 110	0.01 66	0	0.01 35	0.00 36
20%	0.02 47	0.03 78	0.00 58	0.04 31	0.00 20	0.05 50	0.01 56
30%	0.05 14	0.07 76	0.01 16	0.10 06	0.00 22	0.11 00	0.02 80
40%	0.09 54	0.12 85	0.02 21	0.15 77	0.00 25	0.17 45	0.04 95
50%	0.13 83	0.16 73	0.03 27	0.22 70	0.00 45	0.26 45	0.06 87
60%	0.18 73	0.21 72	0.04 26	0.30 21	0.00 66	0.34 18	0.08 88
70%	0.23 72	0.25 05	0.05 65	0.38 31	0.00 91	0.41 18	0.10 48
80%	0.27 37	0.28 53	0.07 25	0.46 75	0.01 40	0.47 21	0.11 85
90%	0.31 62	0.31 00	0.08 31	0.54 17	0.01 73	0.51 52	0.13 05
100%	0.35 23	0.32 21	0.09 54	0.60 70	0.02 77	0.54 17	0.13 54

III. Simulation Result and Analysis

a) Simulation Result of Phase Space Portrait

i. *Simulation Results for normal patient*

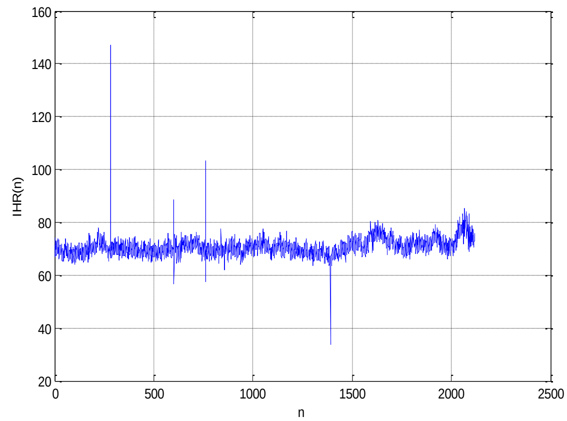

Figure 3.1 : n vs RR interval

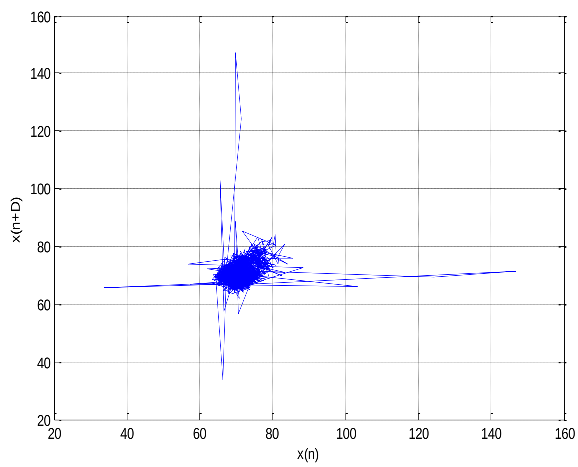

Figure 3.2 : X(n) vs X(n-1)

ii. *Simulation Results for abnormal patient*

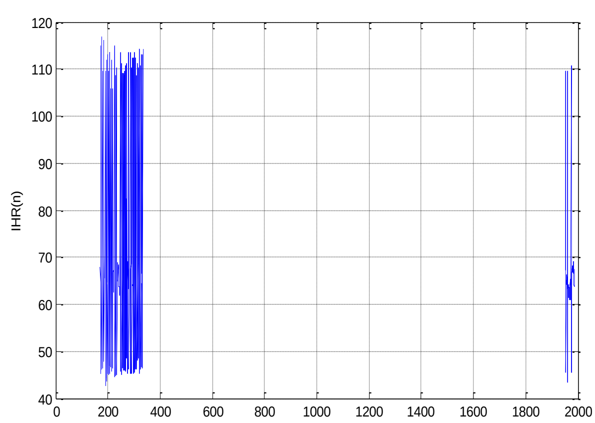

Figure 3.3 : n vs RR interval

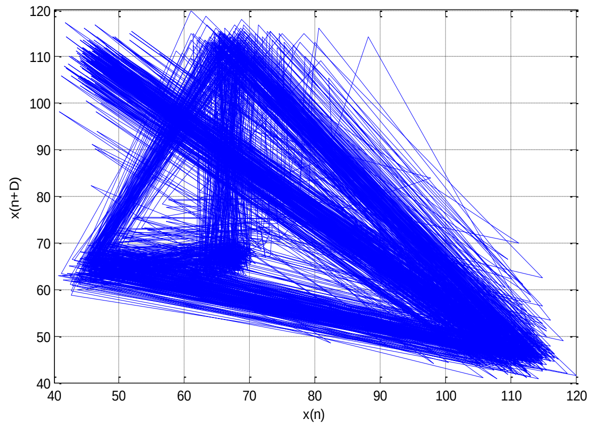

Figure 3.4 : X(n) vs X(n-1)

From the phase space plot for IHR, there lies significant difference between normal and abnormal rhythms. For the normal rhythm, there is normal attractor which forms a slope of almost 45 degree with the axes and there is slight dispersion around that attractor. For the abnormal rhythm, it is seen that their phase space portrait fill more space in the plane and there is random attractor present in the plot.

b) Simulation Result of Poincare plot Analysis

i. *Comparison of simulation results between normal & abnormal patients*

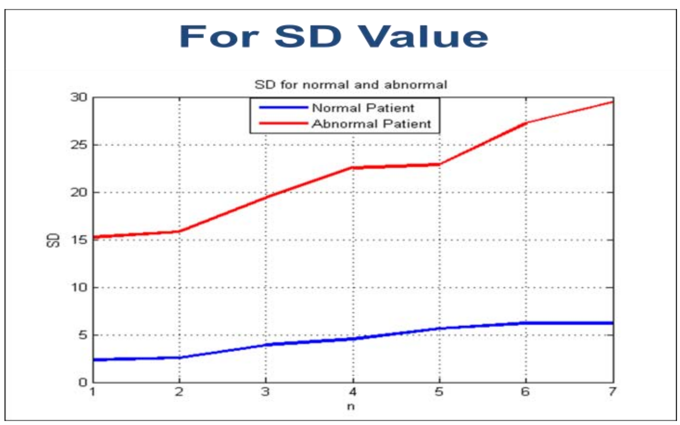

Figure 3.5 : SD Value for normal & abnormal patient

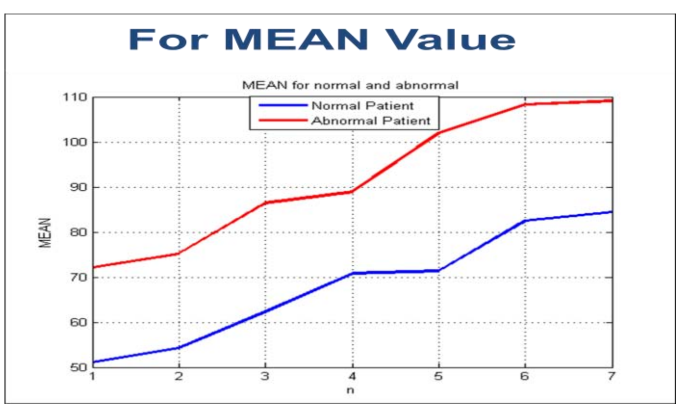

Figure 3.6 : MEAN Value for normal & abnormal patient

Here from the Fig it is seen that the abnormal patients SD is higher than the normal patients. Similarly the abnormal patients MEAN VALUE, VAREANCE, RMSSD, SDSD, is more than the normal patients

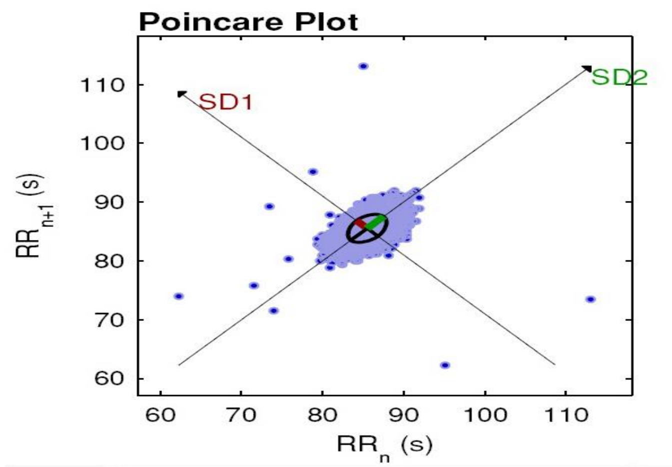

Figure 3.7 : Poincare plot for normal patient [16]

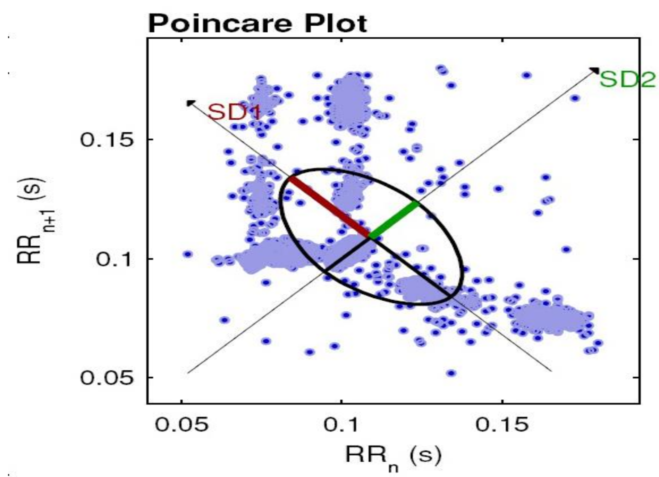

Figure 3.8 : Poincare plot for abnormal patient [16].

Here from the figure it is seen that the all data of the normal patients is close to center. But for the abnormal patients the data scatter from the center as a result the area fill-up by the abnormal patients is more than the area fill-up by the normal patients.

ii. *Comparison based on Poincare plot*

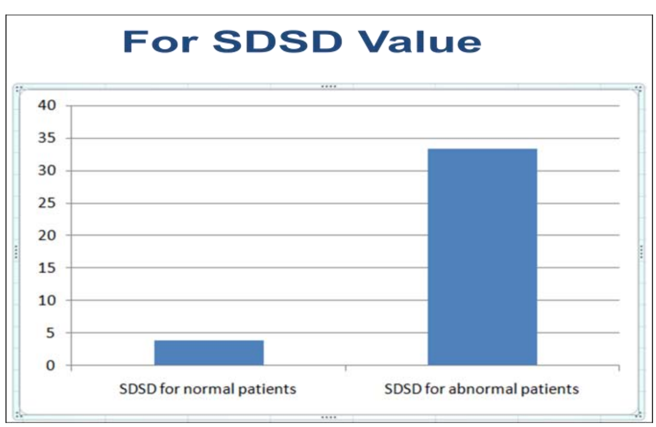

Figure 3.9 : SDSD for normal & abnormal patient

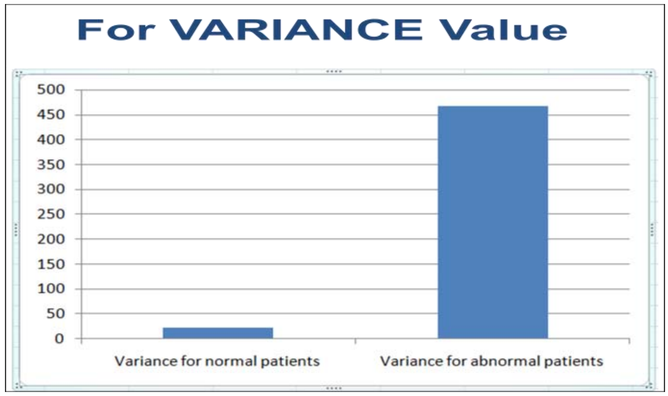

Figure 3.10 : VARIANCE for normal & abnormal patient

From the above figure is seen that the SDSD VALUE for normal patients is 3.92 and for abnormal patients is 33.47.Theoretically for normal patient value of SDSD should be less than 5 and for abnormal patient value of SD should be less than 5.where correct value is achieved and clearly detect the abnormal patients. Similarly for VARIANCE perfect value is achieved to detect the normal and abnormal patients.

c) Simulation Result of Detrended Fluctuation Analysis

i. *Comparison between normal and abnormal patients for DFA*

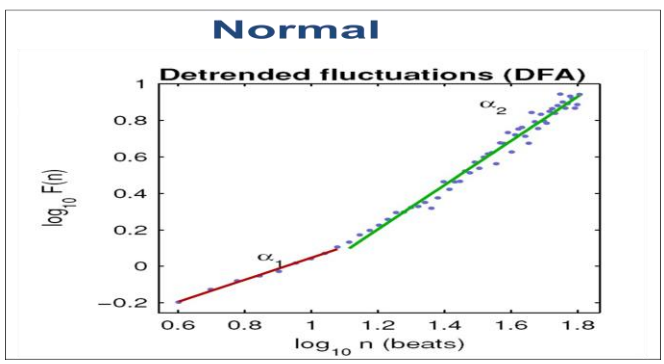

Figure 3.11 : DFA for normal patients

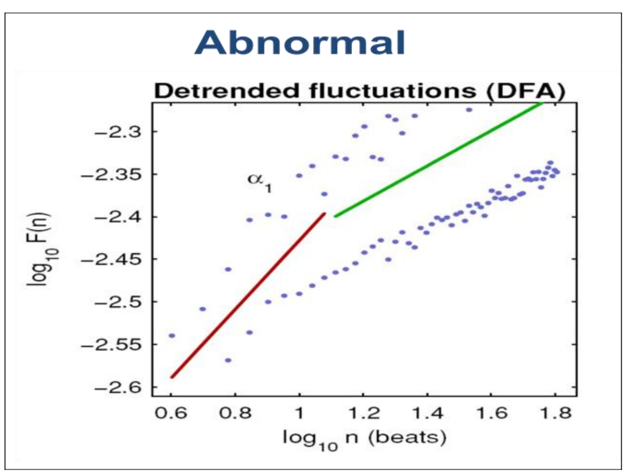

Figure 3.12 : DFA for abnormal patients

Using the DFA method it can be distinguished healthy from unhealthy subjects. Also can be determined which signal is more regular and less complex – useful for analyzing biomedical signals. It's concluded that using non- linear dynamics methods like DFA method is a quantitatively and qualitatively study of physiological signals.

ii. *Comparison based on AREA for Detrended Fluctuation Analysis*

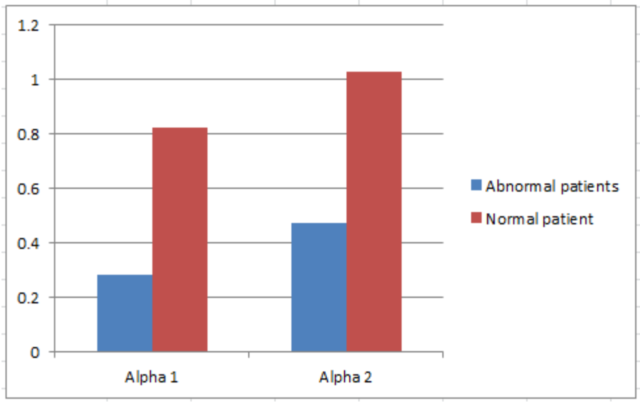

Figure 3.13 : Comparison of AREA between normal & abnormal patients.

From Fig it is seen that Alpha1&2 for normal patients is more than the abnormal patients.

So, if here compare the four techniques it is used namely phase space portrait, Poincare plot DFA and the central tendency measure with the following facts should came out . Phase space portrait only gives us a visual observation of the ECG signals, whether they are from normal or abnormal rhythms. Poincare plot & DFA is more complex to find the normal and abnormal rhythms. On the contrary, central tendency measure quantifies the abnormality levels present in the ECG signals. Moreover, it roughly gives an idea about the abnormality type as observed in our work.

d) Simulation Result of Central Tendency Measure

i. *Simulation Result for both SD-4 and SD-6*

CTM value for SD 10% :

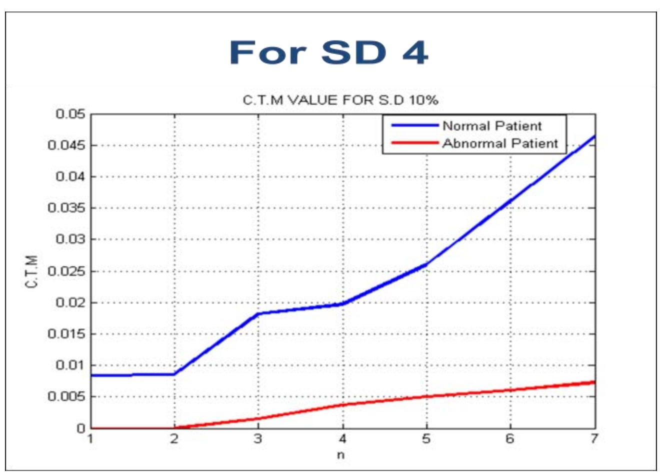

Figure 3.14 : CTM for normal & abnormal patient

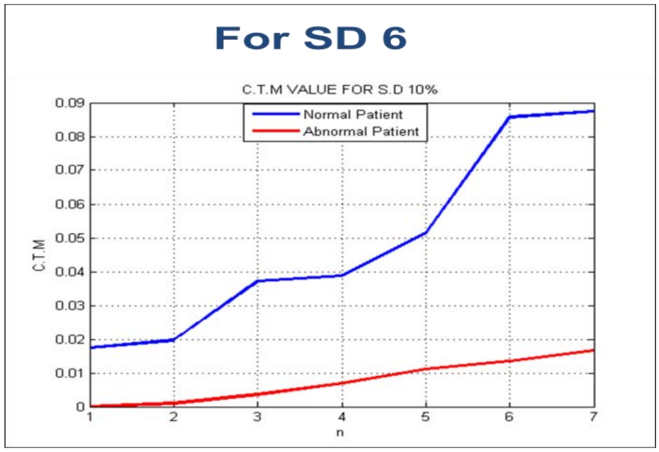

Figure 3.15 : CTM for normal & abnormal patient

CTM value for SD 100% :

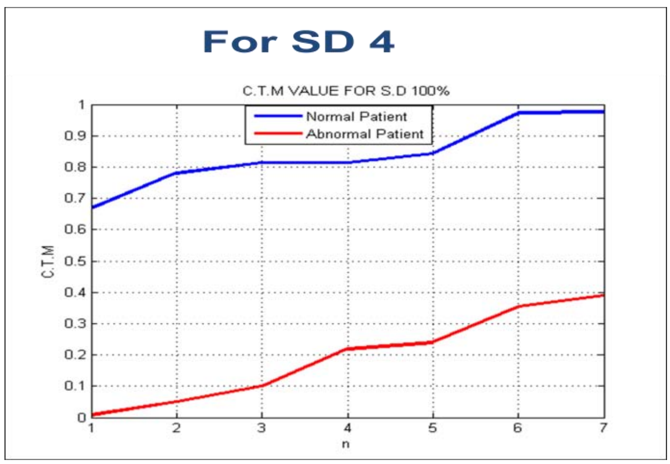

Figure 3.16 : CTM for normal & abnormal patient

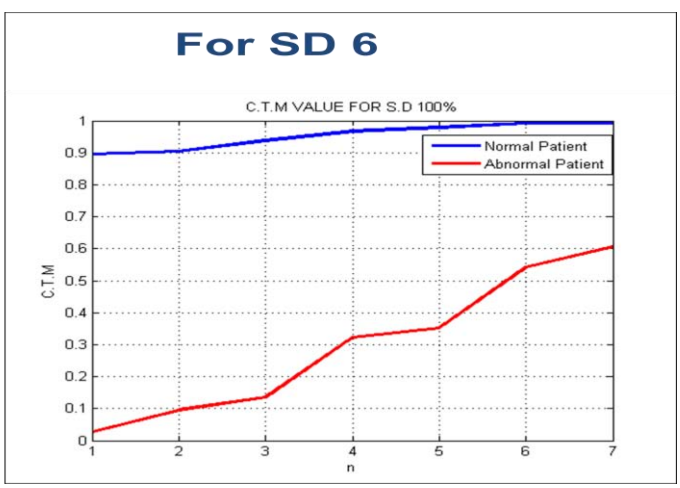

Figure 3.17 : CTM for normal & abnormal patient

Average CTM form 10% to 100% :

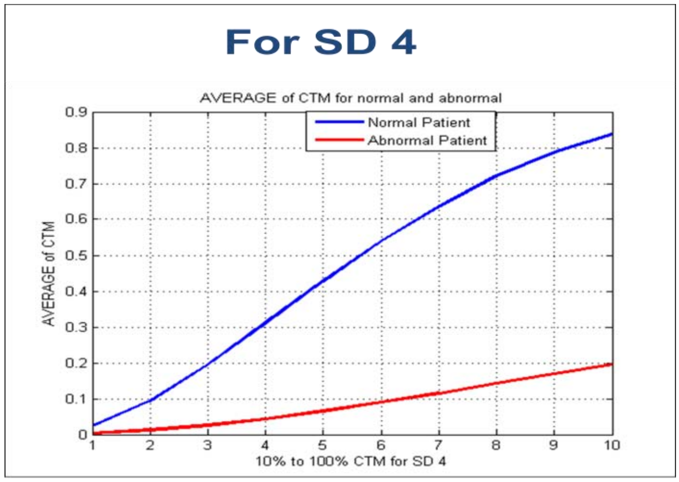

Figure 3.18 : Avg CTM for normal & abnormal patient

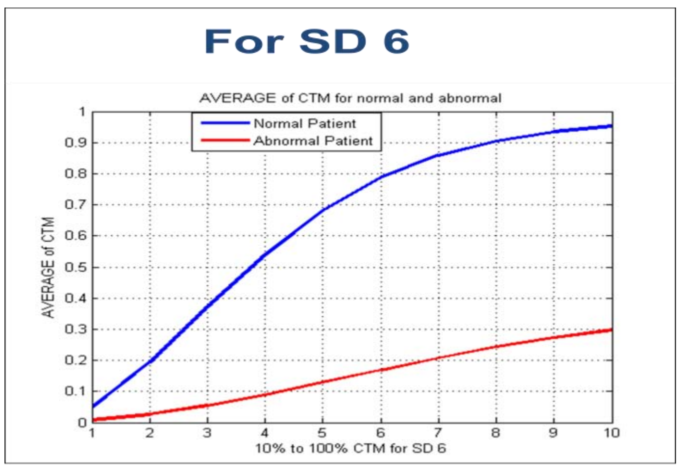

Figure 3.19 : Avg CTM for normal & abnormal patient

Here from the Figure 3.14 to 3.19 it is seen that central tendency is gradually increased with respect to the standard deviation than the normal patients. At the same way for the abnormal patients central tendency is not sharply increased with respect to standard deviation and the CTM values is always lower than 0.5 for abnormal patients. The normal patients's CTM value is similarly increased with respect to SD increase from 10% to 100 %. But for abnormal patients CTM values is increased gradually with respect to SD increase from 10% to 100. The normal patients's CTM value's is much higher than both abnormal patients's so it can be perfectly said that the normal patients's is much more healthy than other normal patients.

ii. *Comparison based on CTM*

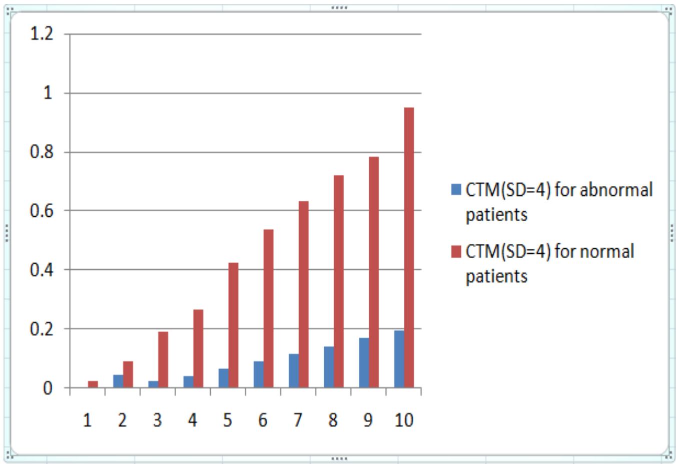

Figure 3.20 : Comparison between normal & abnormal patients (SD-4)

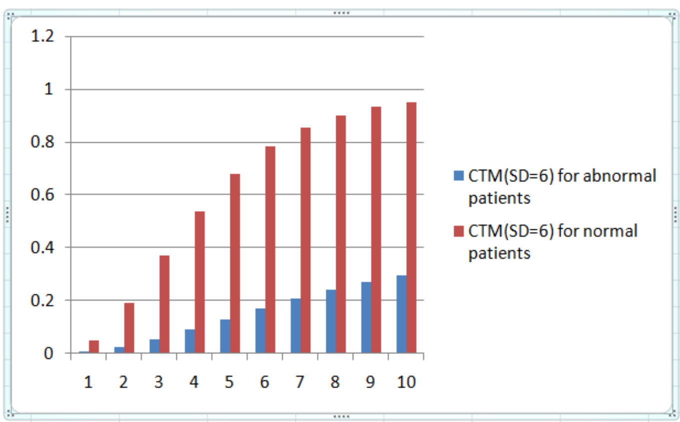

Figure 3.21 : Comparison between normal & abnormal patients (SD-6)

From this Figure 3.20 when the SD = 4 then the average value of CTM for normal patients is more than the CTM of abnormal patients. From this Fig 3.20 it is found that when the SD = 6 then the average value of CTM for normal patients is more than the CTM of abnormal patients. But when SD=6 is used it provides more clear data than SD = 4.

So, here comparing the four techniques namely phase space portrait, Poincare plot DFA and the central tendency measure, it can come out with the following facts. Phase space portrait only gives us a visual observation of the ECG signals, whether they are from normal or abnormal rhythms. Poincare plot & DFA is more complex to find the normal and abnormal rhythms. On the contrary, central tendency measure quantifies the abnormality level presented in the ECG signals. CTM is simpler and give a clear idea than the other techniques. Moreover it roughly gives an idea about the abnormality type as observed in this work.

IV. Conclusion and Future Work

In this work PVC in ECG data set have been identified. The whole work is based on the fact that R-R intervals for normal rhythm data set tend to invariant and for the abnormal rhythm data set tend to vary a lot. This work describes the application of phase space portrait, Poincare Plot, DFA and CTM. Phase Space Portrait is a visible technique. From Poincare Plot a significant difference between normal and abnormal rhythm have been achieved. DFA determine the fluctuation of RR interval from the Slope. For normal rhythm value of CTM is more than the abnormal rhythm. Here clear difference for the normal and abnormal rhythm and high level of accuracy between them has been achieved. So it can be said that it is better to use CTM for classifying the ECG as normal or abnormal. In this paper abnormality of ECG signal have been detected specially in PVC cases. In future several frequency domain methods (i.e cross entropy analysis, Lyapunov Exponents, Support Vector Machine (SVM), Discrete Cosine Transform (DCT)) will be added to detect the abnormalities of heart. Future work may also include working with more number of abnormal records to generalize the detection of beat abnormality type.

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Fellows

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- The "FARSE" is a dignified title which is accorded to a person's name viz. Dr. John E. Hall, Ph.D., FARSE or William Walldroff, M.S., FARSE.

FARSE accrediting is an honor. It authenticates your research activities. After recognition as FARSE, you can add 'FARSE' title with your name as you use this recognition as additional suffix to your status. This will definitely enhance and add more value and repute to your name. You may use it on your professional Counseling Materials such as CV, Resume, and Visiting Card etc.

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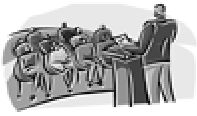

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We shall provide you intimation regarding launching of e-version of journal of your stream time to time.This may be utilized in your library for the enrichment of knowledge of your students as well as it can also be helpful for the concerned faculty members.

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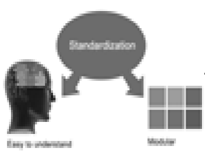

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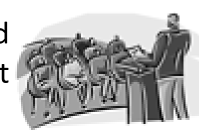

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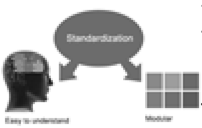

After nomination of your institution as "Institutional Fellow" and constantly functioning successfully for one year, we can consider giving recognition to your institute to function as Regional/Zonal office on our behalf.
The board can also take up the additional allied activities for betterment after our consultation.

The following entitlements are applicable to individual Fellows:

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Note :

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4. Manuscript's Category,

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Before start writing a good quality Computer Science Research Paper, let us first understand what is Computer Science Research Paper? So, Computer Science Research Paper is the paper which is written by professionals or scientists who are associated to Computer Science and Information Technology, or doing research study in these areas. If you are novel to this field then you can consult about this field from your supervisor or guide.

TECHNIQUES FOR WRITING A GOOD QUALITY RESEARCH PAPER:

1. Choosing the topic: In most cases, the topic is searched by the interest of author but it can be also suggested by the guides. You can have several topics and then you can judge that in which topic or subject you are finding yourself most comfortable. This can be done by asking several questions to yourself, like Will I be able to carry our search in this area? Will I find all necessary recourses to accomplish the search? Will I be able to find all information in this field area? If the answer of these types of questions will be "Yes" then you can choose that topic. In most of the cases, you may have to conduct the surveys and have to visit several places because this field is related to Computer Science and Information Technology. Also, you may have to do a lot of work to find all rise and falls regarding the various data of that subject. Sometimes, detailed information plays a vital role, instead of short information.

2. Evaluators are human: First thing to remember that evaluators are also human being. They are not only meant for rejecting a paper. They are here to evaluate your paper. So, present your Best.

3. Think Like Evaluators: If you are in a confusion or getting demotivated that your paper will be accepted by evaluators or not, then think and try to evaluate your paper like an Evaluator. Try to understand that what an evaluator wants in your research paper and automatically you will have your answer.

4. Make blueprints of paper: The outline is the plan or framework that will help you to arrange your thoughts. It will make your paper logical. But remember that all points of your outline must be related to the topic you have chosen.

5. Ask your Guides: If you are having any difficulty in your research, then do not hesitate to share your difficulty to your guide (if you have any). They will surely help you out and resolve your doubts. If you can't clarify what exactly you require for your work then ask the supervisor to help you with the alternative. He might also provide you the list of essential readings.

6. Use of computer is recommended: As you are doing research in the field of Computer Science, then this point is quite obvious.

7. Use right software: Always use good quality software packages. If you are not capable to judge good software then you can lose quality of your paper unknowingly. There are various software programs available to help you, which you can get through Internet.

8. Use the Internet for help: An excellent start for your paper can be by using the Google. It is an excellent search engine, where you can have your doubts resolved. You may also read some answers for the frequent question how to write my research paper or find model research paper. From the internet library you can download books. If you have all required books make important reading selecting and analyzing the specified information. Then put together research paper sketch out.

9. Use and get big pictures: Always use encyclopedias, Wikipedia to get pictures so that you can go into the depth.

10. Bookmarks are useful: When you read any book or magazine, you generally use bookmarks, right! It is a good habit, which helps to not to lose your continuity. You should always use bookmarks while searching on Internet also, which will make your search easier.

11. Revise what you wrote: When you write anything, always read it, summarize it and then finalize it.

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15. Use of direct quotes: When you do research relevant to literature, history or current affairs then use of quotes become essential but if study is relevant to science then use of quotes is not preferable.

16. Use proper verb tense: Use proper verb tenses in your paper. Use past tense, to present those events that happened. Use present tense to indicate events that are going on. Use future tense to indicate future happening events. Use of improper and wrong tenses will confuse the evaluator. Avoid the sentences that are incomplete.

17. Never use online paper: If you are getting any paper on Internet, then never use it as your research paper because it might be possible that evaluator has already seen it or maybe it is outdated version.

18. Pick a good study spot: To do your research studies always try to pick a spot, which is quiet. Every spot is not for studies. Spot that suits you choose it and proceed further.

19. Know what you know: Always try to know, what you know by making objectives. Else, you will be confused and cannot achieve your target.

20. Use good quality grammar: Always use a good quality grammar and use words that will throw positive impact on evaluator. Use of good quality grammar does not mean to use tough words, that for each word the evaluator has to go through dictionary. Do not start sentence with a conjunction. Do not fragment sentences. Eliminate one-word sentences. Ignore passive voice. Do not ever use a big word when a diminutive one would suffice. Verbs have to be in agreement with their subjects. Prepositions are not expressions to finish sentences with. It is incorrect to ever divide an infinitive. Avoid clichés like the disease. Also, always shun irritating alliteration. Use language that is simple and straight forward. put together a neat summary.

21. Arrangement of information: Each section of the main body should start with an opening sentence and there should be a changeover at the end of the section. Give only valid and powerful arguments to your topic. You may also maintain your arguments with records.

22. Never start in last minute: Always start at right time and give enough time to research work. Leaving everything to the last minute will degrade your paper and spoil your work.

23. Multitasking in research is not good: Doing several things at the same time proves bad habit in case of research activity. Research is an area, where everything has a particular time slot. Divide your research work in parts and do particular part in particular time slot.

24. Never copy others' work: Never copy others' work and give it your name because if evaluator has seen it anywhere you will be in trouble.

25. Take proper rest and food: No matter how many hours you spend for your research activity, if you are not taking care of your health then all your efforts will be in vain. For a quality research, study is must, and this can be done by taking proper rest and food.

26. Go for seminars: Attend seminars if the topic is relevant to your research area. Utilize all your resources.

27. Refresh your mind after intervals: Try to give rest to your mind by listening to soft music or by sleeping in intervals. This will also improve your memory.

28. Make colleagues: Always try to make colleagues. No matter how sharper or intelligent you are, if you make colleagues you can have several ideas, which will be helpful for your research.

29. Think technically: Always think technically. If anything happens, then search its reasons, its benefits, and demerits.

30. Think and then print: When you will go to print your paper, notice that tables are not be split, headings are not detached from their descriptions, and page sequence is maintained.

31. Adding unnecessary information: Do not add unnecessary information, like, I have used MS Excel to draw graph. Do not add irrelevant and inappropriate material. These all will create superfluous. Foreign terminology and phrases are not apropos. One should NEVER take a broad view. Analogy in script is like feathers on a snake. Not at all use a large word when a very small one would be sufficient. Use words properly, regardless of how others use them. Remove quotations. Puns are for kids, not grunt readers. Amplification is a billion times of inferior quality than sarcasm.

32. Never oversimplify everything: To add material in your research paper, never go for oversimplification. This will definitely irritate the evaluator. Be more or less specific. Also too, by no means, ever use rhythmic redundancies. Contractions aren't essential and shouldn't be there used. Comparisons are as terrible as clichés. Give up ampersands and abbreviations, and so on. Remove commas, that are, not necessary. Parenthetical words however should be together with this in commas. Understatement is all the time the complete best way to put onward earth-shaking thoughts. Give a detailed literary review.

33. Report concluded results: Use concluded results. From raw data, filter the results and then conclude your studies based on measurements and observations taken. Significant figures and appropriate number of decimal places should be used. Parenthetical remarks are prohibitive. Proofread carefully at final stage. In the end give outline to your arguments. Spot out perspectives of further study of this subject. Justify your conclusion by at the bottom of them with sufficient justifications and examples.

34. After conclusion: Once you have concluded your research, the next most important step is to present your findings. Presentation is extremely important as it is the definite medium though which your research is going to be in print to the rest of the crowd. Care should be taken to categorize your thoughts well and present them in a logical and neat manner. A good quality research paper format is essential because it serves to highlight your research paper and bring to light all necessary aspects in your research.

Informal Guidelines of Research Paper Writing

Key points to remember:

- Submit all work in its final form.
- Write your paper in the form, which is presented in the guidelines using the template.
- Please note the criterion for grading the final paper by peer-reviewers.

Final Points:

A purpose of organizing a research paper is to let people to interpret your effort selectively. The journal requires the following sections, submitted in the order listed, each section to start on a new page.

The introduction will be compiled from reference matter and will reflect the design processes or outline of basis that direct you to make study. As you will carry out the process of study, the method and process section will be constructed as like that. The result segment will show related statistics in nearly sequential order and will direct the reviewers next to the similar intellectual paths throughout the data that you took to carry out your study. The discussion section will provide understanding of the data and projections as to the implication of the results. The use of good quality references all through the paper will give the effort trustworthiness by representing an alertness of prior workings.

Writing a research paper is not an easy job no matter how trouble-free the actual research or concept. Practice, excellent preparation, and controlled record keeping are the only means to make straightforward the progression.

General style:

Specific editorial column necessities for compliance of a manuscript will always take over from directions in these general guidelines.

To make a paper clear

· Adhere to recommended page limits

Mistakes to evade

- Insertion a title at the foot of a page with the subsequent text on the next page
- Separating a table/chart or figure - impound each figure/table to a single page
- Submitting a manuscript with pages out of sequence

In every sections of your document

· Use standard writing style including articles ("a", "the," etc.)

· Keep on paying attention on the research topic of the paper

· Use paragraphs to split each significant point (excluding for the abstract)

· Align the primary line of each section

· Present your points in sound order

· Use present tense to report well accepted

· Use past tense to describe specific results

· Shun familiar wording, don't address the reviewer directly, and don't use slang, slang language, or superlatives

· Shun use of extra pictures - include only those figures essential to presenting results

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Choose a revealing title. It should be short. It should not have non-standard acronyms or abbreviations. It should not exceed two printed lines. It should include the name(s) and address (es) of all authors.

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The summary should be two hundred words or less. It should briefly and clearly explain the key findings reported in the manuscript--must have precise statistics. It should not have abnormal acronyms or abbreviations. It should be logical in itself. Shun citing references at this point.

An abstract is a brief distinct paragraph summary of finished work or work in development. In a minute or less a reviewer can be taught the foundation behind the study, common approach to the problem, relevant results, and significant conclusions or new questions.

Write your summary when your paper is completed because how can you write the summary of anything which is not yet written? Wealth of terminology is very essential in abstract. Yet, use comprehensive sentences and do not let go readability for briefness. You can maintain it succinct by phrasing sentences so that they provide more than lone rationale. The author can at this moment go straight to shortening the outcome. Sum up the study, with the subsequent elements in any summary. Try to maintain the initial two items to no more than one ruling each.

- Reason of the study - theory, overall issue, purpose
- Fundamental goal
- To the point depiction of the research
- Consequences, including definite statistics - if the consequences are quantitative in nature, account quantitative data; results of any numerical analysis should be reported
- Significant conclusions or questions that track from the research(es)

Approach:

- Single section, and succinct
- As a outline of job done, it is always written in past tense
- A conceptual should situate on its own, and not submit to any other part of the paper such as a form or table
- Center on shortening results - bound background information to a verdict or two, if completely necessary
- What you account in an conceptual must be regular with what you reported in the manuscript
- Exact spelling, clearness of sentences and phrases, and appropriate reporting of quantities (proper units, important statistics) are just as significant in an abstract as they are anywhere else

Introduction:

The **Introduction** should "introduce" the manuscript. The reviewer should be presented with sufficient background information to be capable to comprehend and calculate the purpose of your study without having to submit to other works. The basis for the study should be offered. Give most important references but shun difficult to make a comprehensive appraisal of the topic. In the introduction, describe the problem visibly. If the problem is not acknowledged in a logical, reasonable way, the reviewer will have no attention in your result. Speak in common terms about techniques used to explain the problem, if needed, but do not present any particulars about the protocols here. Following approach can create a valuable beginning:

- Explain the value (significance) of the study
- Shield the model - why did you employ this particular system or method? What is its compensation? You strength remark on its appropriateness from a abstract point of vision as well as point out sensible reasons for using it.
- Present a justification. Status your particular theory (es) or aim(s), and describe the logic that led you to choose them.
- Very for a short time explain the tentative propose and how it skilled the declared objectives.

Approach:

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- Sort out your thoughts; manufacture one key point with every section. If you make the four points listed above, you will need a least of four paragraphs.

- Present surroundings information only as desirable in order hold up a situation. The reviewer does not desire to read the whole thing you know about a topic.
- Shape the theory/purpose specifically - do not take a broad view.
- As always, give awareness to spelling, simplicity and correctness of sentences and phrases.

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This part is supposed to be the easiest to carve if you have good skills. A sound written Procedures segment allows a capable scientist to replacement your results. Present precise information about your supplies. The suppliers and clarity of reagents can be helpful bits of information. Present methods in sequential order but linked methodologies can be grouped as a segment. Be concise when relating the protocols. Attempt for the least amount of information that would permit another capable scientist to spare your outcome but be cautious that vital information is integrated. The use of subheadings is suggested and ought to be synchronized with the results section. When a technique is used that has been well described in another object, mention the specific item describing a way but draw the basic principle while stating the situation. The purpose is to text all particular resources and broad procedures, so that another person may use some or all of the methods in one more study or referee the scientific value of your work. It is not to be a step by step report of the whole thing you did, nor is a methods section a set of orders.

Materials:

- Explain materials individually only if the study is so complex that it saves liberty this way.
- Embrace particular materials, and any tools or provisions that are not frequently found in laboratories.
- Do not take in frequently found.
- If use of a definite type of tools.
- Materials may be reported in a part section or else they may be recognized along with your measures.

Methods:

- Report the method (not particulars of each process that engaged the same methodology)
- Describe the method entirely
- To be succinct, present methods under headings dedicated to specific dealings or groups of measures
- Simplify - details how procedures were completed not how they were exclusively performed on a particular day.
- If well known procedures were used, account the procedure by name, possibly with reference, and that's all.

Approach:

- It is embarrassed or not possible to use vigorous voice when documenting methods with no using first person, which would focus the reviewer's interest on the researcher rather than the job. As a result when script up the methods most authors use third person passive voice.
- Use standard style in this and in every other part of the paper - avoid familiar lists, and use full sentences.

What to keep away from

- Resources and methods are not a set of information.
- Skip all descriptive information and surroundings - save it for the argument.
- Leave out information that is immaterial to a third party.

Results:

The principle of a results segment is to present and demonstrate your conclusion. Create this part a entirely objective details of the outcome, and save all understanding for the discussion.

The page length of this segment is set by the sum and types of data to be reported. Carry on to be to the point, by means of statistics and tables, if suitable, to present consequences most efficiently.You must obviously differentiate material that would usually be incorporated in a study editorial from any unprocessed data or additional appendix matter that would not be available. In fact, such matter should not be submitted at all except requested by the instructor.

Content

- Sum up your conclusion in text and demonstrate them, if suitable, with figures and tables.
- In manuscript, explain each of your consequences, point the reader to remarks that are most appropriate.
- Present a background, such as by describing the question that was addressed by creation an exacting study.
- Explain results of control experiments and comprise remarks that are not accessible in a prescribed figure or table, if appropriate.
- Examine your data, then prepare the analyzed (transformed) data in the form of a figure (graph), table, or in manuscript form.

What to stay away from

- Do not discuss or infer your outcome, report surroundings information, or try to explain anything.
- Not at all, take in raw data or intermediate calculations in a research manuscript.
- Do not present the similar data more than once.
- Manuscript should complement any figures or tables, not duplicate the identical information.
- Never confuse figures with tables - there is a difference.

Approach

- As forever, use past tense when you submit to your results, and put the whole thing in a reasonable order.
- Put figures and tables, appropriately numbered, in order at the end of the report
- If you desire, you may place your figures and tables properly within the text of your results part.

Figures and tables

- If you put figures and tables at the end of the details, make certain that they are visibly distinguished from any attach appendix materials, such as raw facts
- Despite of position, each figure must be numbered one after the other and complete with subtitle
- In spite of position, each table must be titled, numbered one after the other and complete with heading
- All figure and table must be adequately complete that it could situate on its own, divide from text

Discussion:

The Discussion is expected the trickiest segment to write and describe. A lot of papers submitted for journal are discarded based on problems with the Discussion. There is no head of state for how long a argument should be. Position your understanding of the outcome visibly to lead the reviewer through your conclusions, and then finish the paper with a summing up of the implication of the study. The purpose here is to offer an understanding of your results and hold up for all of your conclusions, using facts from your research and generally accepted information, if suitable. The implication of result should be visibly described. Infer your data in the conversation in suitable depth. This means that when you clarify an observable fact you must explain mechanisms that may account for the observation. If your results vary from your prospect, make clear why that may have happened. If your results agree, then explain the theory that the proof supported. It is never suitable to just state that the data approved with prospect, and let it drop at that.

- Make a decision if each premise is supported, discarded, or if you cannot make a conclusion with assurance. Do not just dismiss a study or part of a study as "uncertain."
- Research papers are not acknowledged if the work is imperfect. Draw what conclusions you can based upon the results that you have, and take care of the study as a finished work
- You may propose future guidelines, such as how the experiment might be personalized to accomplish a new idea.
- Give details all of your remarks as much as possible, focus on mechanisms.
- Make a decision if the tentative design sufficiently addressed the theory, and whether or not it was correctly restricted.
- Try to present substitute explanations if sensible alternatives be present.
- One research will not counter an overall question, so maintain the large picture in mind, where do you go next? The best studies unlock new avenues of study. What questions remain?
- Recommendations for detailed papers will offer supplementary suggestions.

Approach:

- When you refer to information, differentiate data generated by your own studies from available information
- Submit to work done by specific persons (including you) in past tense.
- Submit to generally acknowledged facts and main beliefs in present tense.

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Topics	Grades		
	A-B	**C-D**	**E-F**
Abstract	Clear and concise with appropriate content, Correct format. 200 words or below	Unclear summary and no specific data, Incorrect form Above 200 words	No specific data with ambiguous information Above 250 words
Introduction	Containing all background details with clear goal and appropriate details, flow specification, no grammar and spelling mistake, well organized sentence and paragraph, reference cited	Unclear and confusing data, appropriate format, grammar and spelling errors with unorganized matter	Out of place depth and content, hazy format
Methods and Procedures	Clear and to the point with well arranged paragraph, precision and accuracy of facts and figures, well organized subheads	Difficult to comprehend with embarrassed text, too much explanation but completed	Incorrect and unorganized structure with hazy meaning
Result	Well organized, Clear and specific, Correct units with precision, correct data, well structuring of paragraph, no grammar and spelling mistake	Complete and embarrassed text, difficult to comprehend	Irregular format with wrong facts and figures
Discussion	Well organized, meaningful specification, sound conclusion, logical and concise explanation, highly structured paragraph reference cited	Wordy, unclear conclusion, spurious	Conclusion is not cited, unorganized, difficult to comprehend
References	Complete and correct format, well organized	Beside the point, Incomplete	Wrong format and structuring

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